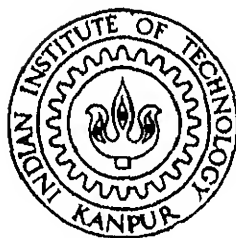


Study and Design of MPEG-2 Transport Stream Multiplexer of Private Data Broadcast

by
D V S N Murty



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Department of Electrical Engineering
INDIAN INSTITUTE OF TECHNOLOGY KANPUR
MAY 1998

Study and Design of MPEG 2 Transport Stream Multiplexer for Private Data Broadcast

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Master of Technology

by
D V S N Murty

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**DEPARTMENT OF ELECTRICAL ENGINEERING
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Certificate

This is to certify that the work contained in this thesis entitled *Study and Design of MPEG-2 Transport Stream Multiplexer for Private Data Broadcast* by D.V.S.N. Murty has been carried out under my supervision and that this work has not been submitted elsewhere for a degree.



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To
my pedanannagaru (uncle)
Sri D V Siva Rao

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D V S N MURTY

Abstract

Digital TV and audio broadcasts are poised to replace the current analog TV broadcast for entertainment news and other one way broadcast applications. Broadcast channels provide high quality with good SNR at affordable costs. Digital TV provides two new opportunities besides current broadcast services: first emerging services based on video on demand and second data broadcast related applications. Data broadcasting from a central server site with low bandwidth interactive channels to clients provides new broadcast applications in the form of hybrid network to bring Internet and Electronic commerce to homes at affordable cost.

In this thesis a review of MPEG 2 standard for video, audio, and private data compression and multiplexing is first carried out. Then a detailed study on the issues involved in the embedding of private data over the MPEG 2 Transport Stream multiplexing is presented. Finally, some measurements of the proposed private data multiplexing in the Transport Stream are presented based on trial implementation over a software MPEG 2 Transport Stream multiplexer.

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Glossary

- 1 **access unit** An MPEG 2 compressed picture or audio
- 2 **bit rate** The rate at which the compressed bit stream is delivered from the channel to the input of a decoder
- 3 **channel** A digital medium that stores or transports an ISO/IEC 13818 stream
- 4 **constant bit rate** Operation where the bit rate is constant from start to finish of the compressed bit stream
- 5 **elementary stream** A generic term for one of the coded video coded audio or other coded bit stream in PES packets. One elementary stream is carried in a sequence of PES packets with one and only one stream id
- 6 **packet identifier PID** A unique integer value used to associate elementary streams of a program in a single or multi program Transport Stream
- 7 **PES Stream** A PES stream consists of PES packets all of whose payloads consist of data from a single elementary stream and all of which have the same stream_id
- 8 **Private data** Non video and non audio data that can be multiplexed along with video and audio data
- 9 **presentation unit** A decoded Audio Access unit or a decoded picture
- 10 **program** A program is a collection of program elements may be elementary streams. Program elements need not have any defined time base those that do have a common time base are intended for synchronized presentation
- 11 **Program Clock Reference (PCR)** A time stamp in the Transport Stream from which decoder timing is derived
- 12 **Program element** A generic term for one of the elementary streams or other data streams that may have included in a program
- 13 **Program Specific Information (PSI)** PSI consists of normative data which is necessary for the demultiplexing of Transport Streams and the successful regeneration of programs
- 14 **Program Stream** One or more elementary streams that share a common time base combined into a single stream

- 15 **reserved** The term **reserved** when used in the clauses defining the coded bit stream indicates that the value may be used in the future for ISO defined extensions. Unless otherwise specified, all reserved bits shall be set to 1.
- 16 **system target decoder (STD)** A hypothetical reference model of a decoding process used to describe the semantics of an ISO/IEC 13818 multiplexed bit stream.

Chapter 1

Introduction

1.1 Convergence of Different Media

The LAN technologies and Internet have brought communications and computing together and offer us several networked services. Common among these services are *email*, *ftp*, *WWW* and *telnet*. Because of the expansion of Internet, several multimedia and real time applications on Internet such as Java applets based MMI (Multimedia Information) browsing, Internet phone, video conferencing etc. are emerging. Internet is also growing fast as a medium for electronic commerce providing interactive shopping, financial services etc.

At the same time the existing telecom structure is not able to support the increasing demands from the emerging technologies and is switching over to newer technologies. Asynchronous Transfer Mode (ATM), Integrated Services Digital Network (ISDN) and Asymmetrical Digital Subscriber Line (ADSL) are some of the technologies which offer higher bandwidths, higher speeds, symmetric or asymmetric channels and are suitable for applications such as video telephony where not only voice but also picture information need to be transferred over the channels.

On the other hand, Terrestrial Broadcasting which has served analog television for many decades is also becoming digital. The digitization is being led by the projects like Digital Video Broadcasting (DVB) in Europe and Grand Alliance (GA) in USA. The aim of these projects is to increase the number of channels available to the viewers by many folds. These projects have adopted MPEG-2 (Moving Pictures

Experts Group 2) as its standard for compression of video and audio data and for multiplexing the compressed audio and video data streams

Efforts are also going on in a new kind of television called Interactive Television (ITV). This is a new form of digital consumer multimedia service that can give viewers much greater control over the content of the program than is possible with conventional analog television. The aim of this kind of television is to provide services like video on demand, multimedia information retrieval, distance education, home shopping, and multi-player multi-location video games and so forth.

Digital broadcasting technology and Internet are leading to the merging of the computers and the TVs into a new system, mean for entertainment and work at the same time.

1.2 Need to Transmit Private Data over Broadcast Channels

The convergence of the Internet with TV will lead to a concept called hybrid network. Most of the applications of the Interactive TV have higher BW requirements in one direction and a much lower BW required in the reverse direction. For example, in case of the video on demand or digital library access, subscribers send a request to the multimedia server through either ordinary telephone lines or some other low BW links and in the reverse direction large files are downloaded from the server as shown in the Fig. 1.1.

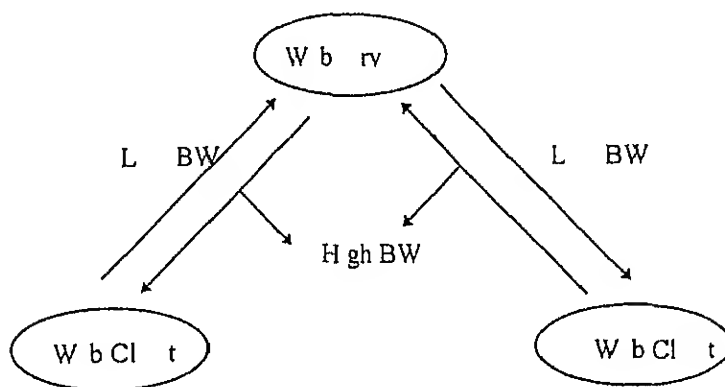


Fig. 1.1 Asymmetric BW requirement of Web based applications

This large bandwidth required in the reverse direction can be achieved by dedicated lines but this would be costly and wasteful as well. As an alternative, existing TV

broadcast channels can be used to transmit these large volumes of data with a significant decrease in the cost when compared to dedicated links

1.3 How MPEG 2 Supports Private Data Broadcast?

MPEG 2 as described in later chapters has a provision to multiplex video, audio, and private data into a single stream called Transport Stream (TS) and transmit that stream through either point-to-point environments like broadcast channels with minimum transmission errors. The international standard ISO/IEC 13818-1 (systems) defines the syntax for multiplexing MPEG 2 compressed audio, video, and private data and to convey the synchronization of audio/video signals [1].

Objective

The objective of this thesis is to study and design the Transport Stream multiplexing and to see how private data can be sent over the broadcast channels along with the ordinary television programs. Here the word *private* has been used in a broad sense. HTML documents, IP packets, multimedia information, teletext are some of the forms of private data. The aim is to build a data broadcasting service over the hybrid network using MPEG 2 at the transport level.

Work done in this area

Most of the work done in this area is under the proprietary rights of the concerned organizations. Interactive TV, Web TV, Internet phone, video on demand are some of the commercial services already available in the U.S., Europe, and Japan. Products such as IRD (Integrated Receiver Decoder), Transport Stream multiplexer, etc. are available in the form of VLSI circuits. But being proprietary, it is useful only for commercial purposes. At the academic level, two students have done some significant work in this area. One is the thesis work done by Mr. Christos Tryfonas of University of California titled "MPEG 2 Transport over ATM Networks" [6]. The other is the thesis work done by Mr. Luigi Atzori of University of Cagliari (Italy) titled "Conveying Multimedia Services within the Transport Stream MPEG 2".

1.4 Organization of this Thesis

This thesis is organized into 5 chapters. The second chapter presents an overview of MPEG compression techniques and different kinds of multiplexing strategies used in MPEG 2 to transmit compressed data along with the private data. In Chapter 3 a detailed description of Transport Stream multiplexing has been presented. Chapter 4 concentrates on implementation issues of a Transport Stream multiplexer and the results we obtained in our studies. Chapter 5 concludes the thesis with the conclusions of our work and the scope of the future work.

Chapter 2

MPEG 2 A Standard for Audio, Video, and Private Data

2.1 Introduction

Digital compression techniques have been traditionally used to reduce the amount of data storage capacity needed to save large files in computers. For the past decade, the global Telecom industry has also been using compression techniques to reduce the bandwidth and consequently the cost required for providing narrow band telephone circuits. Since late 80s communication engineers began developing high capacity very large scale integrated circuits (VLSICs) and sophisticated software routines that could compress high bandwidth real time data such as video.

Digital Television and need for compression

A conventional PAL video signal contains 625 lines per individual image or frame. A single frame of conventional analog video is composed of 174,000 picture elements or pixels; more than 4 millions pixels are being sent to the television screen each second. Digital television transmission systems convert the visual and audio information into streams of binary digits or bits. A resolution of 32 bits per pixel are needed to digitize a single color video signal. This results in a data rate of 128 Mbit/s. A 128 Mbit/s digital signal would fill an entire satellite transponder to capacity. To allow multiple digital TV signals to share a single transponder bandwidth, the digital television signals have to use compression techniques with a compression ratio of more than 20:1.

The advantages of the digital representation of a signal against the analog one are

- a) Increase in the signal quality and robustness to noise
- b) Fidelity of regeneration
- c) Efficient storage and retrieval of TV signals for editing and transmission
- d) Random access of stored data
- e) Digital broadcasting technology has enabled the merging of computers and TVs to ether. Users can use the same equipment for both entertainment as well as for work. The same equipment that decodes and shows entertainment programs(video/movie on demand movie listing games) would also become the basis for interactive transactional and information services(distant education programs bank operations remote databases) as well as a tool to access socially useful information (telemedicine news on demand etc). In such a way digital broadcasting is the driving force for both the integration of diverse media and the extension of computing into our daily life

2.2 The Moving Pictures Experts Group

In 1988 the International Standards Organization (ISO) of the Telecommunications Union established the Moving Pictures Experts Group (MPEG) to achieve an internationally recognized world wide standard for the compressed representation of video film graphic text and audio materials. The committee's goals were to develop a relatively simple inexpensive and flexible standard that put most of the complex functions at the transmitter rather than the receiver.

MPEG 1

In 1991 the MPEG 1 standard was introduced to handle the compressed digital representation of non video sources of multimedia data. MPEG 1 could be adapted for the encoding/compression and transmission of video signals as long as the video material was first converted from the original interlaced mode to a progressively scanned format. This standard was aimed for applications such as storage and retrieval of multimedia data on digital storage media with a target bit rate upto about 2Mbit/s.

MPEG 2

Later the scope of the MPEG group was extended to provide appropriate MPEG 2 video and associated audio compression algorithms for a wide range of

audio visual applications at a substantially higher bit rates which were not successfully covered or envisaged by the MPEG 1 standard. Specifically MPEG 2 was given the charter to provide video quality not lower than NTSC/PAL and up to CCIR 601 quality with bit rates targeted between 4 and 10 Mbit/s. Emerging applications such as digital cable TV distribution networked database services asynchronous transfer mode (ATM) digital video tape recorder (VTR) applications and satellite and terrestrial digital broadcasting distribution were seen to benefit from the increased quality expected to result from the emerging MPEG 2 standard. This standard was released in 1994.

The MPEG 1 and MPEG 2 video compression techniques developed and standardized by the MPEG group have developed into important and successful audio and video coding standards world wide with an increasing number of products becoming available in the market. One key factor for the success is the generic structure of the MPEG standards supporting a wide range of applications and application specific parameters. To support the wide range of applications profiles a diversity of input parameters including flexible picture size and frame rate can be specified by the user. Also the MPEG group only standardized the decoder structures and the bit stream formats. Thus the manufacturers could increase the coding efficiency (i.e. video quality at a given bit rate) by developing innovative encoder algorithms even after the standards were finalized.

MPEG 4

Anticipating the rapid convergence of telecommunications, computer and TV/film industries, the MPEG group officially initiated a new MPEG 4 standardization phase in 1994. MPEG 4 standardizes algorithms and tools for coding and flexible representation of audio visual data to meet the challenges of future multimedia applications and applications environments. In particular, MPEG 4 addresses the need for universal accessibility and robustness in error prone environments with interactive functionality, coding of natural and synthetic data as well as high compression efficiency. MPEG 4 video standard targeted a bit rate of 56 Kbit/s for mobile or public switched telephone network (PSTN) video applications and up to 4 Mbit/s for TV/film applications. The release of the MPEG 4 International Standard is targeted for November 1998.

2.3. Digital Video Broadcasting and MPEG-2

The next generation of TV signal broadcast standards is based on digital data compression and digital data transmission. The success of the MPEG-2 standards, led the Broadcast Engineering community to steer the standardisation of Digital TV around MPEG. The aim of the standardisation process was to provide for both higher image quality and better bandwidth utilization than classical analog color TV broadcasting signals such as PAS, NTSC, and SECAM.

In January 1995, the Digital Video Broadcasting (DVB) project originated by the European Broadcasting Union (EBU) has published a set of formal standards which define the new digital video broadcast system. A list of this is given in Appendix C. They specify the following functions: Format selection, Video coding, Audio coding, Transport and Transmission. These DVB standards are the technical basis for implementing digital TV transmission in Europe, Asia, Australia, and many other regions of the world. In the United States, a Grand Alliance has been formed by proponents of digital HDTV systems. The planned system is a digital simulcast with the current analog NTSC transmission. The goal of the Grand Alliance is to introduce an ATV (advanced television) terrestrial transmission standard for the United States. A simplified block diagram of a DVB receiver is shown in the following Fig. 2.1.

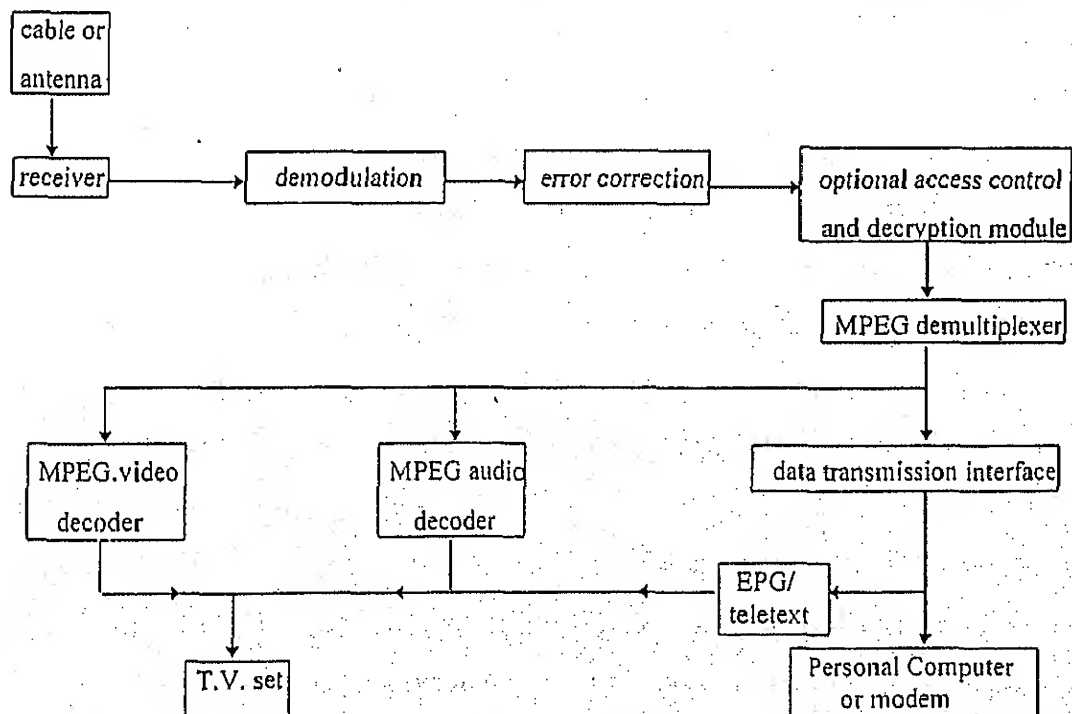


Fig. 2.1 DVB receiver

While the initial focus of the DVB project was delivery of digital video to set top converters these standards are broad enough to cover any form of high speed digital broadcasting including software electronic newspapers Internet Protocol (IP) streams or Web pages Conditional access is one of the main features of the DVB streams The compression technique used by the DVB standard is the MPEG 2 algorithm MPEG 2 is an audio/video compression algorithm optimized for broadcast quality transmission up to HDTV quality The list of MPEG 2 standards (ISO/IEC 13818) is given in Appendix A

2.4 Brief Description of MPEG Compression Techniques

2.4.1 Video Compression

MPEG compression is accomplished by four basic techniques: pre processing, temporal prediction, motion compensation, quantization coding. Pre processing filters out non essential visual information from the video signal information that is difficult to encode but not an important component of human visual perception. Motion compensation takes advantage of the fact that video sequences are most often highly correlated in time. Each frame in any given sequence may be similar to the preceding and the following frame.

The digital video encoder scans subsections within each frame called *macro blocks* and identifies which ones will not change position from one frame to the next. The encoder also identifies predictor macro blocks while noting their position and direction of motion. Only relatively small difference called the *motion compensated residual* between each predictor block and the affected current block is transmitted to the receiver. The receiver decoder stores the information that does not change from frame to frame in its buffer memory and uses it periodically to fill in the blanks.

A mathematical algorithm called the Discrete Cosine Transform (DCT) reorganizes the residual difference between frame from a spatial domain into an equivalent series of coefficient numbers in a frequency domain that can be more quickly transmitted. Quantization coding converts these sets of coefficient numbers into more compact representative numbers. The encoder refers to an internal index of *code book* of possible representative numbers from which it selects the code word that best matches each set of coefficients. Quantization coding also rounds off all

coefficient values, within a certain range of limits, to the same value. Although this results in an approximation of the original signal, it is close enough to the original to be accepted for most viewing applications. A simplified block diagram of MPEG-2 video decoder is shown in the following Fig. 2.2.

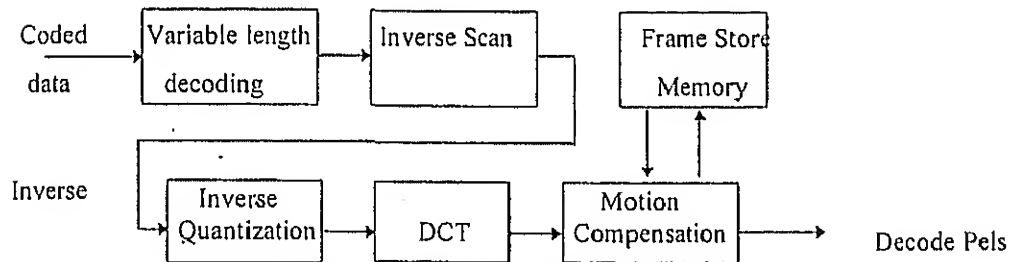


Fig. 2.2 Simplified Video Decoding Process

b) Audio Compression:

The main objective of audio compression is to represent an audio source with as few bits as possible, preserving the level of quality as perceived by the human ear. The ISO/IEC MPEG-2 Layer II encoding uses a sub-band representation of the audio signal in order to take advantage of the frequency masking properties of the human hearing system. The frequency spectrum of the audio signal is separated into sub-bands by the use of a sub-band or transform filter bank. This results in a representation of the audio signal by sub-band samples.

The sub-band signals may be quantized because the resulting quantization noise will be at a similar frequency, and relatively low signal to noise ratios (SNRs) are acceptable due to psychoacoustic phenomenon of masking. A psychoacoustic model of human hearing determines what actual SNR is acceptable in each sub-band. A bit allocation operation distributes the available bits among the sub-bands in accordance with the required SNR. The sub-band values are quantized to the precision indicated by the bit allocation operation and formatted into the audio elementary stream.

The selection of an appropriate audio source coding schemes for different applications like broadcasting and multimedia is of vital importance and should be considered carefully from a number of view points. MPEG defines three coding schemes namely Layer I, Layer II, and Layer III whose complexity increases in the

given order. MPEG-2 adopts Layer II encoding scheme as its audio encoding standard. This is a multichannel audio encoding scheme.

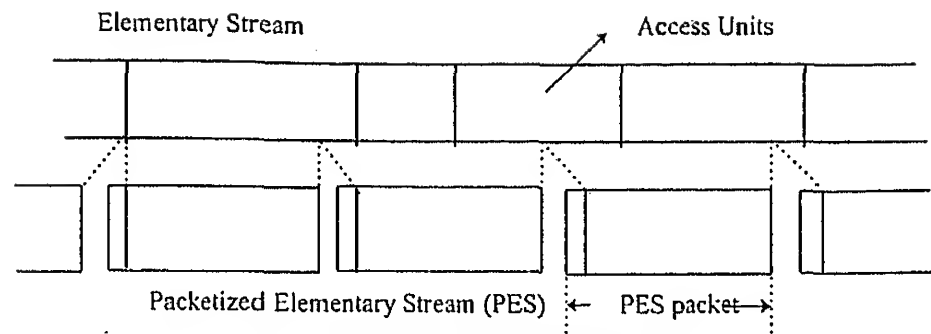
Besides audio and video coding compression, the DVB standards specify the transport mechanism as MPEG-2 Systems coding. The MPEG-2 TS is a method of combining or *multiplexing* multiple video, audio, and private data streams together. The DVB standards then add other specifications, some media independent and some unique to a particular media (like satellite links), to specify the recommended combinations of technical parameters such as modulation and forward error correction coding to insure interoperability of products.

2.5 Multiplexing of Audio, Video, and Private data

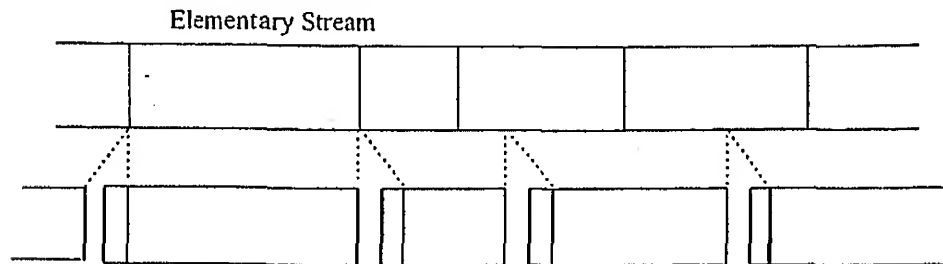
The MPEG-2 systems layer (ISO/IEC 13818-1 [1]) addresses the combining of one or more elementary streams (MPEG-2 compressed video and audio streams) of video and audio, as well as uncompressed private data, into a single or multiple streams which are suitable for storage or transmission. Systems coding follows the syntactical and semantic rules imposed by this specification and provides information to enable synchronized decoding at decoder buffers over a wide range of retrieval or receipt conditions.

Systems coding can be carried out in two different ways: the *Program Stream* and the *TS*.

Program Stream This is a multiplexing approach to multiplex one or more elementary streams (compressed video and audio streams) of a single program (i.e., elementary streams with a common time reference) into a single stream. In the first stage of the multiplexing, the data from elementary streams are packetized as shown in Fig. 2.3. These packets are called *PES (Packetized Elementary Stream) packets*. These PES packets consist of a PES header followed by packet data (which is taken from elementary streams). These PES packets in a second stage are organized in *packs*. A pack consists of a pack header and is followed by a variable number of PES packets. A Program Stream begins with a system header that may optionally be repeated. This system header carries a summary of the system parameters defined in the stream. A detailed description of Program Stream can be found in [1]. The



2.3.a Packetization using fixed size PES packets



2.3.b Packetization using variable size PES packets

Fig.2.3 Generation of a PES from an elementary stream

Program Stream approach is used when requirement for compatibility with MPEG-1 is stipulated.

TS: This is a multiplexing approach to multiplex video, audio elementary streams along with private data streams. In the first stage of the multiplexing the video, audio elementary streams and data streams are packetized. These resulting packets are called *PES (Packetised Elementary Stream) packets*. In the second stage of the multiplexing the PES packets are put in the payload portion of Transport packets. The header portion of the Transport Stream (TS) packets are specified by the ISO/IEC 13818-1.

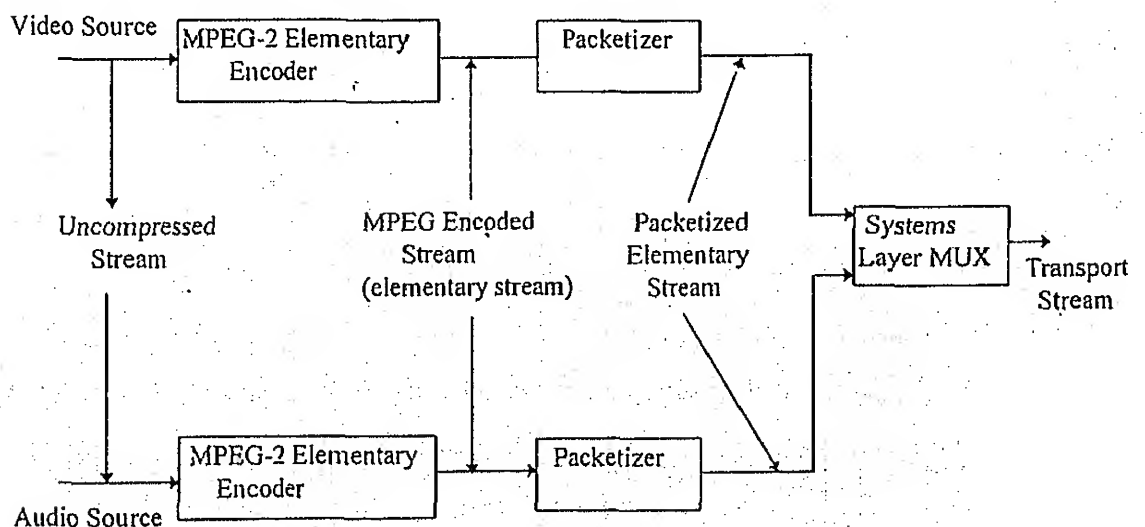


Fig. 2.4 Simplified overview of Systems layer (TS case)

standard [1] Each TS packet consists of 188 bytes out of which 4 bytes are allocated for header and 184 bytes for payload. It combines one or more programs with one or more independent time bases into a single stream. The two different schemes of Systems coding are motivated by different application requirements. For example, TSs are appropriate for environments where errors and data loss events are likely. Program Streams are appropriate for relatively error-free media.

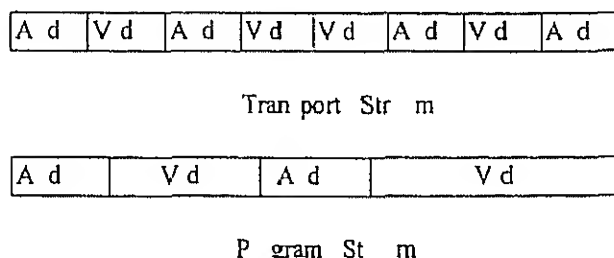


Fig 2.5 Packaging Approach s

The TS is designed in such a way that several operations on a TS are possible with minimum effort. Among these are:

- Extract the TS packets from one program within the TS and produce as output a different TS with only that one program.

- Extract the TS packets of one or more programs from one or more TSs and produce as output a different TS.

- Extract the contents of one program from the TS and produce as output a Program Stream containing that one program.

- Take a Program Stream, convert it into a TS to carry it over a lossy environment, and then recover a valid and in certain cases identical Program Stream.

In the case of Program Stream, none of the above operations are possible. We cannot transmit a Program Stream by enclosing a TS in that.

TSs are appropriate for environments where errors and data loss events are likely, including certain storage media and transmission on noisy channels. Program Streams are appropriate for relatively error-free media such as CD-ROMs.

The DVB project had adopted TS multiplexing as its standard to multiplex the audio, video, and private data streams for the following reasons:

- TSs are intended for multi-program applications such as broadcasting. Program streams cannot support multi-program applications.

TSs can be transmitted in error prone environments like broadcast channels where errors are more likely to occur

TSs are designed to support constant bit rate applications such as video broadcasting

In the following chapter a detailed description of the TS multiplexing will be presented

Chapter 3

Transport Stream Multiplexing

3.1 Introduction

As discussed in chapter 2 Transport Stream (TS) multiplexing is suitable for error prone environments and its use is recommended in distributing compressed bit streams over long distance networks and in broadcast systems. The TS may be constructed from one or more programs (here program is equivalent to a channel in analog television jargon) from elementary streams from Program Streams (PS) or from other TSs which may themselves contain one or more programs.

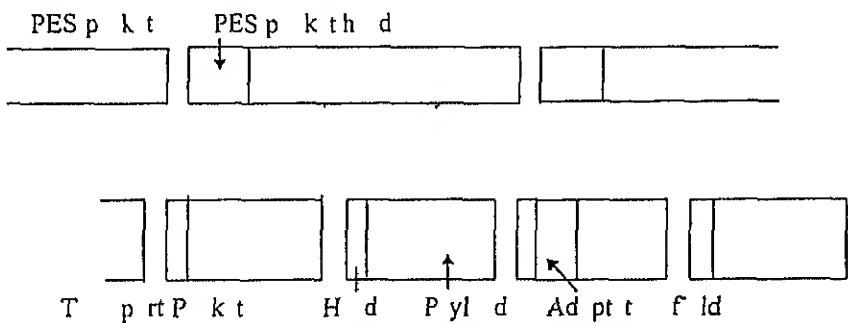


Fig 3.1 TS generation from PES packets

As seen in section 2.5 TSs are constructed from *PES packets* and packets containing other necessary information as shown in Fig. 3.1. The TS supports the multiplexing of video and audio elementary streams from multiple programs with independent time bases. As shown in Fig. 3.1 the first byte of each PES packet header is located at the first available payload location of a TS Packet. The packet size of a TS packet is fixed at 188 bytes. The advantages of using fixed packet length are given in the following section.

3.2 Advantages of using fixed length packets in TSs

The fixed length packetization approach offers flexibility and some additional advantages when multiplexing data related to several applications into a single bit stream.

a) dynamic capacity allocation

While digital systems are generally described as flexible, the use of fixed length packets offers a high level of flexibility to allocate channel capacity among video, audio, and auxiliary data services. The use of *PID* field in the Transport packet header as a means of bit stream identification makes it possible to have a mix of video, audio, and auxiliary data which is flexible and which need not be specified in advance. The entire channel capacity can be reallocated to meet immediate service needs including allocation of the entire bit stream for delivery of data services. This concept is termed as *dynamic capacity allocation*.

b) scalability

The ability to dynamically allocate system capacity may be exploited to allow additional elementary bit streams to be added at the input of the multiplexer or to allow these elementary bit streams to be multiplexed at a second stage with the original bit stream. The presence of multiple elementary bit streams in the data channel allows the system to be scalable.

c) extensibility

The standard for system coding was developed with the understanding that there would be future services that could not be anticipated at the introduction of the service. It was therefore extremely important that the transport architecture be open ended. New elementary bit streams could be handled at the transport layer without hardware modifications by assigning new PIDs at the transmitter and filtering out these new PIDs in the bit stream at the receiver. Backward compatibility was assured when new bit streams were introduced into the transport system since existing decoders would automatically ignore new PIDs. This capability could possibly be used to compatibly introduce newer higher temporal or spatial resolution services or stereoscopic services by sending augmentation data along with the normal television service data. The presence of multiple elementary bit streams in the data channel and

provision for identification of yet unidentified future services allow the system to be extensible

d) error handling and robustness

Another fundamental advantage of the fixed length packetization approach is that the fixed length packet can form the basis for handling errors that occur during transmission. Error correction and detection processing (which precedes packet demultiplexing in the receiver subsystem) may be synchronized to the packet structure so that one deals in the decoder at the packet level when handling data loss due to transmission impairments. Essentially after detecting errors during transmission one recovers the data bit stream from the first good packet. Recovery of synchronization within each application is also aided by the transport packet header information. Without this approach recovery of synchronization in the bit streams would be completely dependent on the properties of each elementary bit stream. The presence of fixed length packets improves the system's robustness.

The next section discusses the details of TS multiplexing in detail.

3.3 Transport Stream Multiplexing and Synchronization

A group of elementary streams (both audio and video) with a common system_clock_frequency time base is called a program. Data from each elementary stream are encoded and multiplexed together with information that allows elementary streams within a program to be replayed in synchronism. The standard ISO/IEC 13818-1 defines a hypothetical decoder known as the *TS System Target Decoder (TSTD)* for this purpose [1]. The TSTD is a conceptual model used to define the byte arrival and decoding events and the times at which these occur precisely and to model the decoding process during the construction or verification of TSs.

3.3.1 TSTD

There are three types of decoders in the TSTD: video, audio, and systems as shown in Fig. 3.2.

Input to the system target decoder is a TS. Even though the TS may contain multiple programs with independent time bases, the TSTD decodes only one program at a time. In the TSTD model all timing indications refer to the time base of that program. Data from the TS enters the TSTD at a piecewise constant rate. The 1^{st}

byte enters at a time $t(i)$. The time at which this byte enters the T-STD can be recovered from the input stream by decoding the input *program_clock_reference* (PCR) field, encoded in the TS packet *adaptation field* of

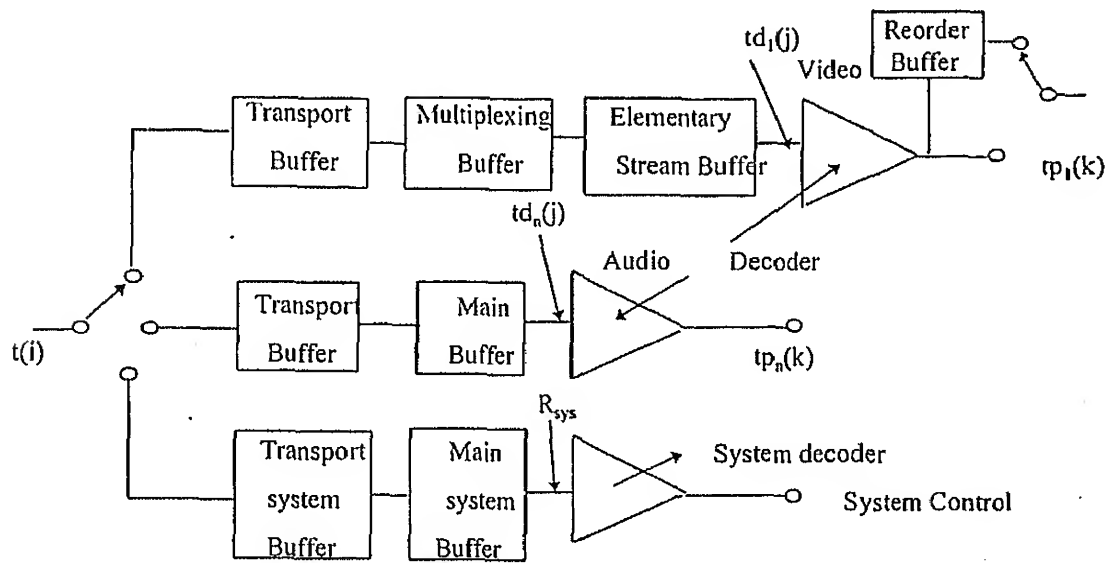


Fig. 3.2 TS system target decoder

the program to be decoded. The PCR field is described in Appendix B. Complete TS packets containing data from an elementary stream n , as indicated by its PID (explained in later sections), are passed to the transport buffer for stream n . This includes duplicate TS packets and packets with no payload.

3.3.2. Timing Model:

The MPEG-2 standard assumes a timing model in which the end-to-end delay from the signal input to the encoder to the signal output of the decoder must be constant. This delay is the sum of encoding, encoder buffering, decoder buffering, decoding, and presentation delays. All the timing information referenced in the T-STD model is defined in terms of a common system clock referred to as the System Time Clock (STC). This timing information is carried by various fields in the PES packet headers and TS packet headers. These fields are described later in Appendix B. The value of the *system_clock_frequency* is chosen as 27MHz. It should meet the following constraints:

$$27\text{MHz} - 810 \leq \text{system_clock_frequency} \leq 27\text{MHz} + 810$$

$$\text{rate of change of system_clock_frequency with time} \leq 75 * 10^{-3} \text{ Hz/s}$$

3.3.3 System Time Clock (STC) recovery in decoder

The PCR values in the TS header are used to implement clock reconstruction loops in decoders with sufficient accuracy as shown in Fig. 3.3. In an ideally constructed and delivered MPEG-2 stream, each PCR arrives at the decoder at precisely the time indicated by the value of that PCR. If the decoder's clock matches exactly that of the encoder, then the decoding and presentation of video and audio will automatically have the same rate as those at the encoder. With matched encoder and decoder clock frequencies, any correct PCR can be used in order to set the value of the decoder's System Time Clock (STC) and from that time onwards the decoder's STC will follow that of the encoder's without the need for further adjustments (with a constant phase difference corresponding to the delay between the encoder and decoder). However, this is not the case in real systems where the delays are not constant. In practice, the decoder's free-running system_clock_frequency will not match that of the encoder. This can result in either buffer overflow or underflow at the decoder. This problem can be solved by synchronizing the decoder's clock to the value found in the PCR field of TS header using a PLL (Phase Locked Loop). A simple PLL is shown in the Fig. 3.3.

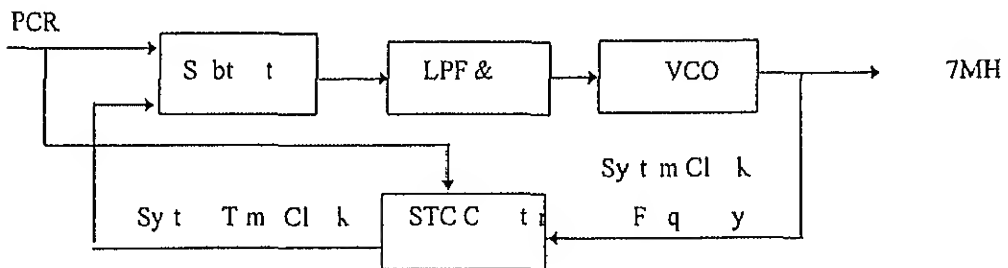


Fig. 3.3 A simple PLL block diagram

The PLL works as follows. Initially, the PLL waits for the arrival of the time base (i.e., the first PCR value received by the program being decoded) and stores this value in the STC counter. The PLL operates in a closed-loop operation. When a new PCR value is received, its value is compared to the local STC. The difference generates the error, which is sent through the low-pass filter and amplifier. This controls the instantaneous frequency of the VCO, whose output provides the decoder's system_clock_frequency. This process continues until the PLL becomes phase-locked, and the error term converges to a constant value, resulting in the variations in the VCO frequency to be zero.

When data streams are to be transported over a communications network the jitter imposed by the network has to be smoothened by de-jittering mechanisms. These involve the use of additional buffering. In case of CBR traffic this results in storage of the packets and play out at a constant rate. But, in the case of the VBR traffic the buffer occupancy varies with the transport rate and additional mechanisms to provide constant average delay through the buffers are needed.

3.3.4 Buffering

All bytes that enter the transport buffer are removed at a certain pre-calculated rate. Bytes which are part of the PES packets or its contents are delivered to the main buffer in the case of audio elementary streams and system data and to the multiplexing buffer for video elementary streams. The header bytes of TS packets are not moved to any buffer but are used for system control. Duplicate TS packets are discarded at the input of the T-STD itself.

Complete TS packets containing system information for the program selected for decoding enter the system transport buffer at the TS rate. These include TS packets whose PID values are 0 or 1 and all TS packets identified via the Program Association Table as having the *program_map_PID* value for the selected program. The transport_rate for the i^{th} byte is given by the following equation:

$$\text{transport_rate}(i) = \frac{(\text{number of bytes between PCRs}) \cdot (\text{system_clock_frequency})}{(\text{PCR} - \text{PCR}_{\text{old}})}$$

The transport buffer size is fixed at 512 bytes. The elementary stream buffer sizes are equal to the *vbv buffer size* as it is carried in the sequence header of a video elementary stream (Refer to ISO/IEC 13818-2 [7]). For the calculation of the elementary buffer size and main buffer size refer to ISO/IEC 13818-1 [1].

In the case of video elementary stream when a data byte is transferred from multiplexing buffer to the elementary buffer all PES packet header bytes that are in multiplexing buffer and immediately preceding that byte are instantaneously removed and discarded. All PES packet payload data bytes enter elementary stream buffer instantaneously upon leaving multiplexing buffer.

For each elementary stream buffer and main buffer all data for the access unit that has been in the buffer for longest duration and any stuffing bytes that immediately

precede it in the buffer at the time $td(j)$ and is removed instantaneously at time $td(j)$ where $td(j)$ is the decoding time for

j^{th} access unit of n^{th} elementary stream. The access units are sent to the decoder instantaneously. The decoding time $td(j)$ is specified in the DTS or PTS fields which are present in the PES packet header portion (explained in the following section). As the access unit is removed it is instantaneously decoded into a presentation unit.

In the case of system data, data is removed from the main buffer at a rate given by the following expression whenever there is at least one byte available in the main buffer:

$$R_y = \max [80000 \text{ bits/s}, \text{transport_rate}(1)/500]$$

Here transport_rate is given by the following expression:

Elementary streams buffered in main buffers and elementary buffers are decoded instantaneously by the decoders. In case of video elementary streams there may be an additional delay in reorder buffers before being presented to the viewer at the output of the T-STD. The time at which a presentation unit is presented to the viewer is $tp(k)$. For presentation units that do not need reordering, the presentation time is equal to the decoding time. The T-STD model presented here ignores the finite and different delays caused by the real audio and visual presentation devices.

Note: The Synchronization issues are presented very briefly in this section. The readers can refer to the standard ISO/IEC 13818-1 for complete details [1].

3.4 Structure of TS Packets

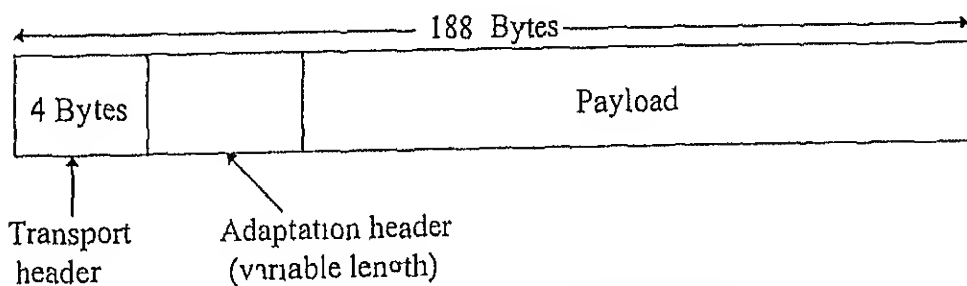


Fig 3.4 Structure of a TS packet

Each TS packet consists of 188 bytes as shown in Fig 3.4. The choice of this packet size is motivated by a few factors. The packet needs to be large enough so that the overhead due to the transport headers does not become a significant portion of the

total data carried. They should also not be so large that the probability of packet error becomes significant under standard operating conditions. It is also desirable to have packet lengths appropriate to the block sizes of typical block oriented error correction approaches so that packets may be synchronized to error correction blocks and the physical layer of the system can aid the packet level synchronization process in the decoder. Another motive for the particular packet length selection is interoperability with the ATM format. This is possible because the ATM cell size is 53 bytes with a 5 byte header and 48 byte payload. So each TS packet is encapsulated in four ATM cells as 47 byte payloads ($4 \times 47 = 188$) leaving space for 1 ATM AAL byte per ATM cell.

The structure of the TS packet header is as shown in Fig 3.5. A detailed description of all the Transport header fields is given in Appendix B.

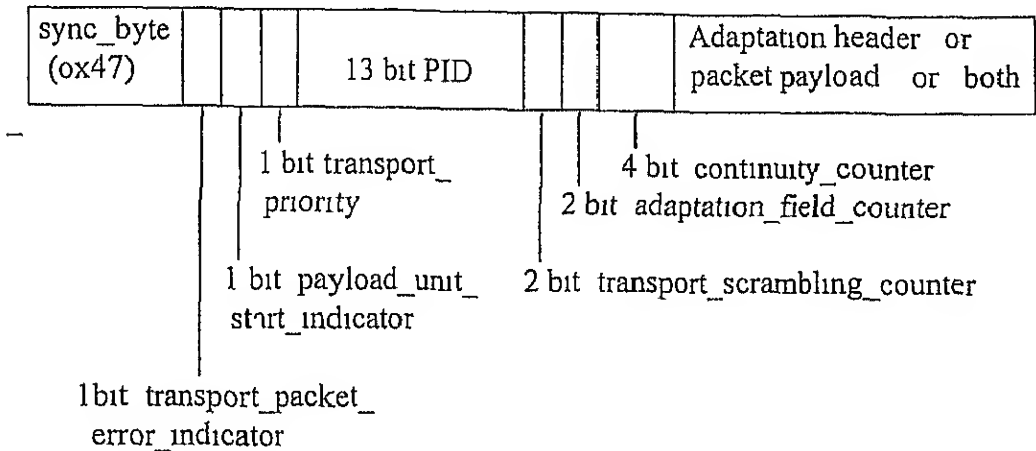


Fig 3.5 Transport Header format

3.4.1 The Adaptation Field

This is a variable length field. Its presence is flagged in the Transport header field with the adaptation_field_control field. This field consists of information useful for higher level decoding functions and uses flags to indicate the presence of particular extensions to the field. The structure of an adaptation field is shown in Fig 3.6. A detailed description of all the fields is given in Appendix B.

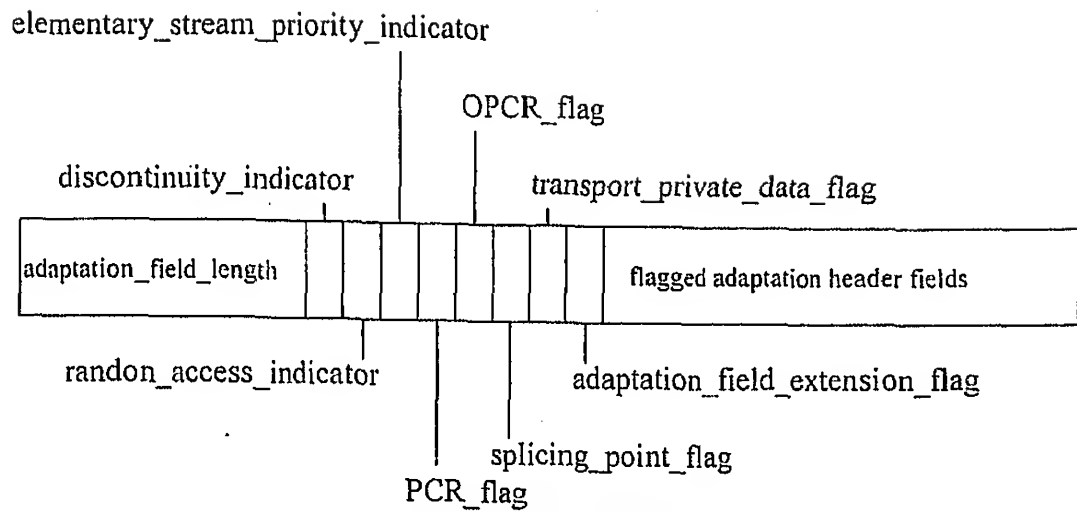


Fig. 3.6 Adaptation Field structure

3.4.2. The PES Packet Format

As stated before, prior to entering the transport layer, some (video and audio) elementary bit streams will go through PES packetization. The PES header carries various rate, timing, and descriptive information as set by the encoder. The PES packet is application dependent resulting in packets of variable length with a maximum definable size of 2^{16} bytes.

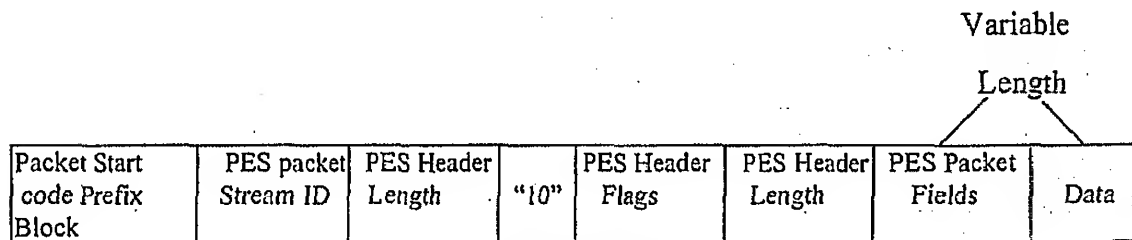


Fig. 3.7 PES packet structure

The PES packet, including the header, are transmitted contiguously as the payload of transport packets. A new PES packet always starts in a new transport packet, and PES packets that end in the middle of a transport packet are followed by *stuffing bytes* for the remaining length of the transport packet. The packet payload is a stream of contiguous bytes of a single elementary stream, and for video or audio packets, the payload is a sequence of access units provided by the encoder corresponding to the video pictures and audio frames. A detailed discussion of the PES packet header fields is given in Appendix B.

3.5 Program Specific Information

The Program Specific Information specifies the data that enable the multiplexing of programs by decoders. Programs are composed of one or more elementary streams each labeled with a PID. Programs or elementary streams or parts thereof may be conditionally accessed. But Program Specific Information (PSI) cannot be scrambled.

In TSs PSI is classified into four table structures as shown in Table 3.1. While these structures may be thought of as simple tables, they shall be segmented into sections and inserted in TS packets.

PSI tables in general are transmitted in the appropriate bit stream sequentially without a gap between the tables. This implies that these tables need not necessarily start at the beginning of a transport packet payload and that therefore there needs to be an indicator as to where these begin in the bit stream. This functionality is achieved with the *pointer_field*. The *pointer_field* is present in the packet if a PSI table begins in the Transport packet. This event is signaled in the Transport packet header by setting the *payload_start_indicator* to 1. The pointer field indicates the number of bytes that

Structure Name	Stream Type	Reserved PID #	Description
Program Association Table	TS	0x00	Associates Program Number and PMT PID
Program Map Table	TS	Assigned	Specifies PID values for components of one or more programs
Network Information Table	Private	Assigned	Physical network parameters such as FDM frequencies, Transporter Numbers, etc.
Conditional Access Table	Private	0x01	Associates one or more (private) EMM streams each with a unique PID value

Table 3.1 Program Specific Information

follow it before the start of a PSI table. As an example, a pointer_field value of 0x00 indicates that a new PSI table begins immediately following it.

3.5.1 Program Association Table

The program association table is transmitted as the payload of the TS packet with PID value 0x0000. These TS packets together shall contain a complete list of all programs within the TSs and describes how program numbers associated with program services map onto TS packets containing program map tables for indicated programs. The program_association_table may be transmitted as multiple program_association_segments with each segment having a maximum length of 1024 bytes. The program_association_table is described in the above table. The transport decoder can extract individual table segments from the bit stream in whatever order it desires. Each table segment has a fixed length 8 byte header component for table segment identification, a variable length component, that depends on the number of entities contained, and 4 byte CRC 32 field as shown in Fig. 8.

Program Association Segment and Table Header Formats

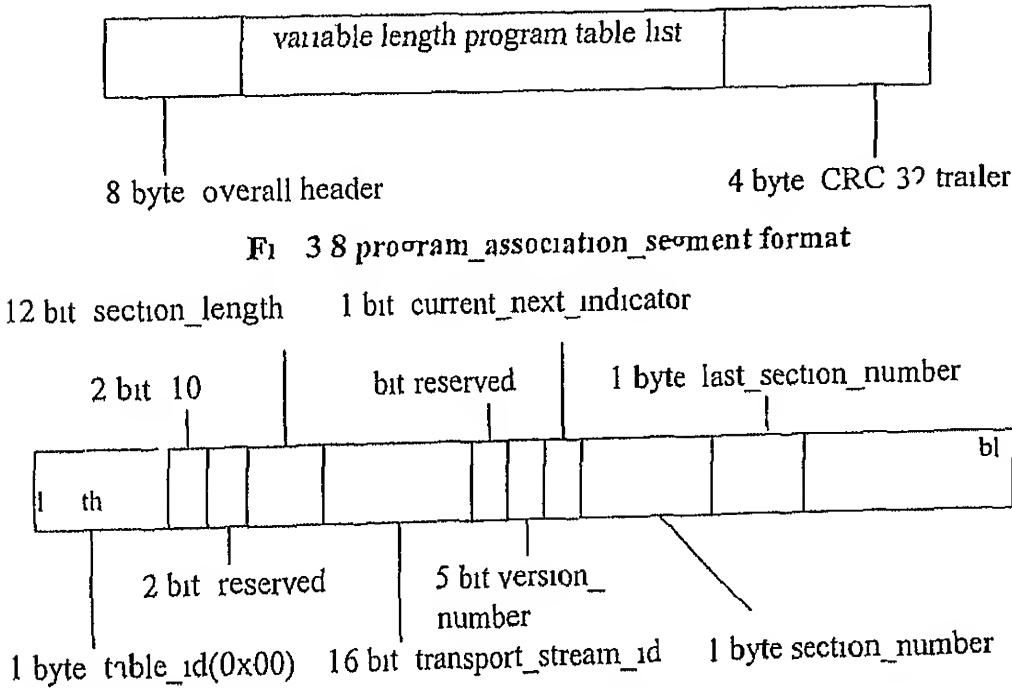


Fig 3.9 program_association_table fixed_length_header

All the fields of the program_association_table are shown in Fig. 9 and are described in Appendix B.

3.5.2 Program Map Table

The program map table provides the mappings between program numbers and the elementary streams that comprise them. A single instance of such a mapping is referred to as a program definition. The program map table is transmitted as the payload of the bit stream with the PID value equal to `program_map_PID` (as indicated in the program association table). This table carries information about the applications that make up programs. Each program map table is transmitted as a single `program_map_section`. The format for a `TS_program_map_section` can be described as a combination of an overall header field, fields that describe each program within the table, and a CRC field as shown in Fig. 3.10. Each `program_map_PID` may contain more than one `program_map_section`, with each one describing a different program.

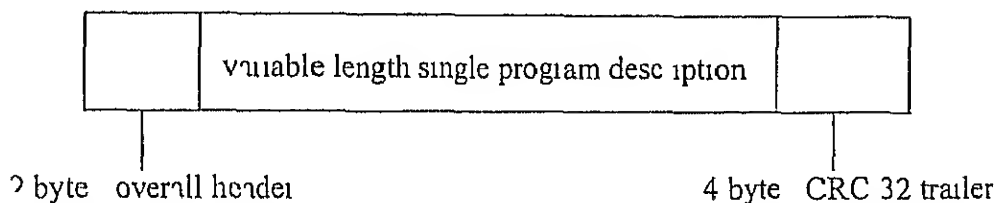


Fig. 3.10 `program_map_section` format

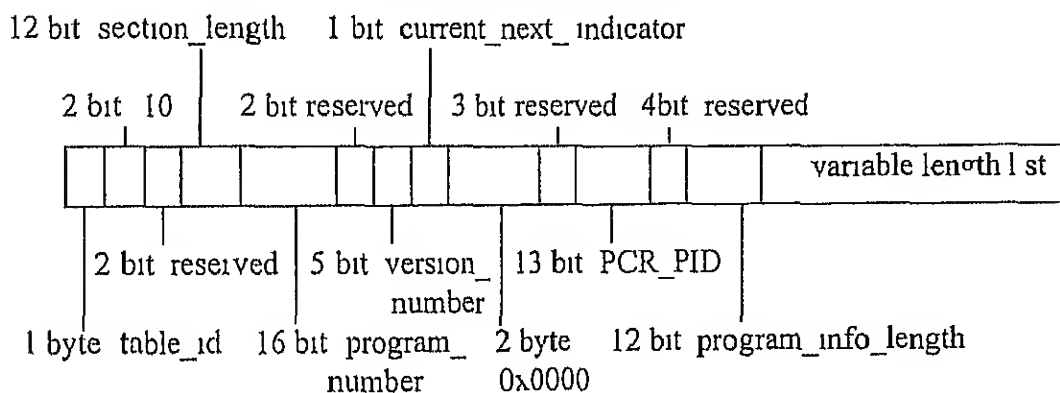


Fig. 3.11 `program_map_section` fixed_length_header

All the fields in the `program_map_section` are shown in Fig. 3.11 and are described in Appendix B.

The `conditional_access_table` is used to scramble the streams at the receiver. `Network_information_table` is used to provide the information about the network in use when the TSs are being sent over a network.

The `conditional_access_table` and `network_information_table` are not discussed in this thesis.

In this chapter and in Appendix *B* a detailed description of the various fields in the TS and PLS packet header is presented. In the following chapter we will discuss the implementation issues involved for a TS multiplexer.

Chapter 4

Implementation Issues of Transport Stream Multiplexing

4.1 Introduction

In the previous chapters a detailed overview of MPEG 2 Systems Layer was presented. This chapter concentrates on the implementation details of a model MPEG 2 TS multiplexer following the specifications of the standard ISO/IEC 13818-1 [1]. As mentioned in the previous chapters the standard gives the designer maximum flexibility in choosing the multiplexing strategy to multiplex different audio, video and private data elementary streams.

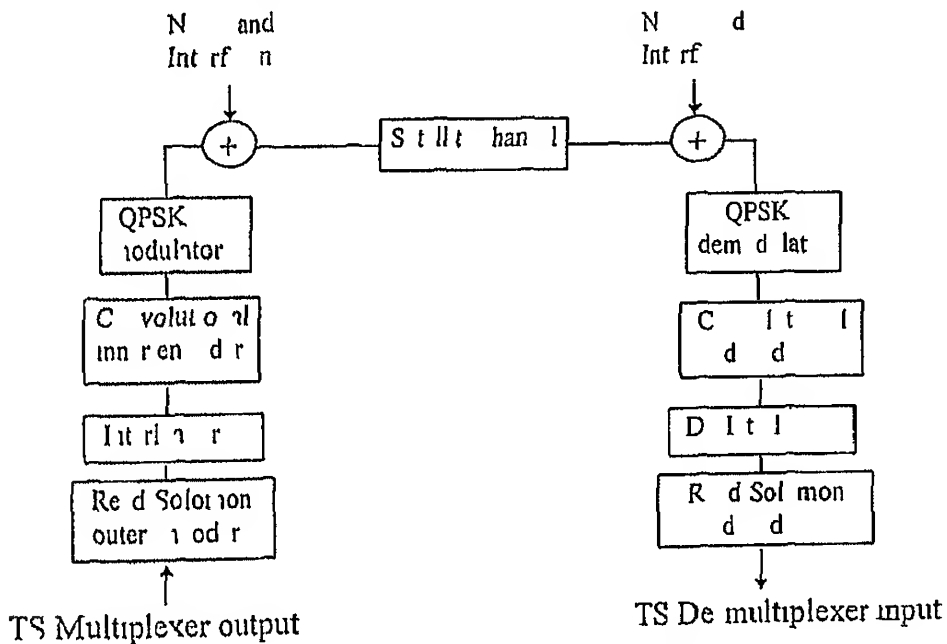


Fig. 4.1 A typical DVB setup

It does not specify any multiplexing strategy and only defines all the fields required to provide enough information to the decoder to smoothly demultiplex and decode the

received TSs. Before transmitting the TS into a digital broadcast channel proper error correction mechanisms should be introduced and modulated as shown in the Fig 4.1. These error correction mechanisms and modulation techniques are beyond the scope of this work.

There are two methods in wide use for multiplexing data from several sources into a single stream. They are *Time Division Multiplexing (TDM)* and *packet multiplexing*. TDM assigns periodic time slots to each of the elementary streams of audio, video and data. The ISDN video conferencing standards (collectively known as H.320) use TDM in their multiplexing standard (H.221). MPEG-1 Systems and MPEG-2 Systems use packet multiplexing.

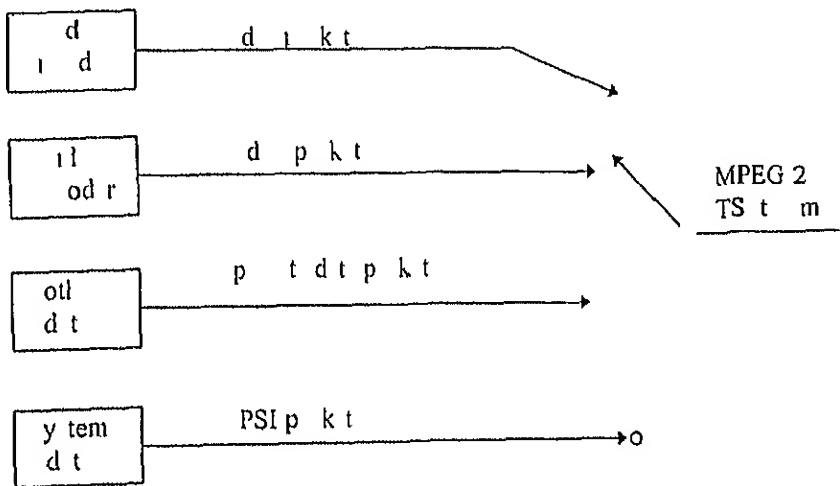


Fig 4.2 Packet multiplexing

In packet multiplexing, data packets from the several elementary streams of audio, video, and data are interleaved one after the other into a single TS as shown in Fig 4.1. Elementary streams and TS can be sent either at constant bit rate (CBR) or at variable bit rates (VBR) simply by varying the lengths or the frequency of the packets appropriately. In particular, elementary streams can be sent as VBR even though the TS itself is transmitted at CBR. This method enables some elementary streams that temporarily do not require very many bits to give up their proportionate channel capacity in favor of other elementary streams that temporarily require more than their share of the overall channel capacity. This functionality is called *statistical multiplexing*.

The TS multiplexing is carried out in two different levels. As mentioned in previous chapters, in the first level the elementary streams are packetized into PES

packets. In the second level the PES packets of all the different streams are multiplexed into TS packets to form a TS. This is nothing but packet multiplexing. We used a multiplexing strategy which will help the decoder to playback the received video and audio streams in synchronism and ensured that transmission of private data takes place only in the unutilized BW of the digital broadcast channel. This is statistical multiplexing.

4.2 Multiplexing Strategy

Since data transmission takes place only in the unutilized BW of the broadcast channels, the multiplexing strategy used by us depends on the output bit rates of MPEG-2 video and audio encoders. We have used MPEG-2 software video and audio decoders available in public domain [14-16] to extract the information such as bit rate, frame rate, resolution, type of picture etc. of sample video and audio elementary streams [13-15]. The inputs to these decoders are the MPEG-2 compressed streams. These decoders provide all the information about the input streams. For example, in the case of video decoder [14], it gives the output bit rate of the encoder that is used to generate the compressed video stream, type of picture that was encoded (PAL or NTSC or SECAM etc.), frame rate and so on. Similarly, in the case of audio decoder [16], it gives information whether the input stream is a Layer I or Layer II or Layer III encoded, output bit rate of the encoder that is used to generate the compressed audio stream, whether the encoded stream is stereo quality or mono quality etc.

From this, we calculated the ratio of the video to audio bit rates of the compressed (elementary) streams. This ratio is nothing but the ratio at which we can interleave the transport packets containing video data and audio data. This kind of interleaving results in perfect synchronism between video and audio streams of a particular program at the decoder. The output bit rate of a TS multiplexer is fixed by considering the following issues:

a) Type of video being transmitted in the TS

The output bit rate of any TS multiplexer depends on the type of video material that is being encoded in it. In practice, MPEG-2 video encoder located at the transmitter side has a finite time to make encoding decisions. Pre-recorded

movies and other taped material do not push the time constraint of the encoder to the limit. The encoder can select at its leisure the most efficient method for encoding at the lowest possible data rate. So for these cases the output bit rate of TS multiplexer is lower.

Type of Video Service	Data Rate (in Mbit/s)
Movies (VHS quality)	1.152
News or Entertainment	4.56
Live Sport Event	4.608
16:9 Wide Screen TV	5.760
Studio Quality Broadcast TV	8.064
HDTV	14.00

Table 4.1 Data rates for different video services

But in the case of live sports and other live action materials a higher data rate is required because the encoder is forced to make immediate coding decisions and must also transmit complex rapid motion changes without introducing high levels of distortion at the decoder's end. Typical output bit rates are given in the above table.

b) Bandwidth of the satellite transponder

In DVB, the total bit stream passing through a single satellite transponder may consist of as many as eight TV services with associated audio, auxiliary audio services, conditional access data, and private data services. The informational bit rate for this transmission may be as high as 49 Mbit/s over a 36 MHz satellite transponder. This implies that for a single TV channel (video+audio+private data) the information bit rate can be approximately 6 MHz.

c) Buffer management in T-STD

In the T-STD model presented in the previous chapter (Fig. 3.2), the transport buffers for different kinds of elementary streams are drained at different rates as shown in the following table.

Elementary Stream	Transport buffer rate
Low level video	4.8 Mbit/s
Main level video	18 Mbit/s
High 1440 level video	72 Mbit/s
High level video	96 Mbit/s
Audio	2 Mbit/s
Systems	1 Mbit/s

Table 4.2 Transport buffer rates for different elementary streams

Video Input buffer overflow and underflow [2]

So while choosing the TS output bit rate we should guarantee that systems transport buffer will neither overflow nor underflow (underflow is intentionally allowed in certain special cases). The input to the transport buffer for systems data is nothing but the TS packets containing PSI information (PAT, PMT, CAT and NIT). Let us assume that in a TS there are y packets containing PSI information. For neither overflow nor underflow to occur in systems transport buffer, the TS bit rate is chosen satisfying the following inequality

$$\left| (y * 188 * 8 * 10^6 / r) - y \right| < 4096$$

where r is the TS rate in Mbit/s. 4096 is the size of the transport buffer in bits. The higher of the two values of r obtained by solving the above inequality is taken as the TS rate.

To satisfy all the above mentioned issues, we have chosen the output bit rate of the TS multiplexer as 6 Mbit/s. This rate is equal to 750 Kbytes/s. We know from previous chapters that the size of a transport packet is 188 bytes. So there will be a total of $\lfloor 750 * 10^3 / 188 \rfloor$ number of packets per second in the TS. Among these we can interleave the video and audio transport packets according to the ratio of the bit rates of the MPEG-2 video and audio streams.

Model Calculations

As an example, let us consider a video stream and audio stream

Name of the video stream *hhilong.m2v*

Name of the audio stream *genzmer2.m2a*

When subjecting these two streams to the respective decoders the following information is obtained

For h264 m2v

<i>aspect ratio</i>	<i>16 9</i>
<i>frame rate</i>	<i>25 fps</i>
<i>bit rate</i>	<i>15 Mbit/s</i>
<i>vbv buffer size</i>	<i>1 7 6704 Mbits</i>
<i>progressive sequence</i>	<i>interlaced</i>
<i>Level</i>	<i>Main</i>

For gsm2 m2a

<i>Layer</i>	<i>II</i>
<i>Total bit rate</i>	<i>384 Kbit/s</i>
<i>Sampling freq</i>	<i>48 Khz</i>
<i>Mode</i>	<i>Stereo</i>

Now using the information we can calculate the ratio at which video and audio transport packets can be interleaved in the TS as the ratio of $(15 \cdot 10^6 / 384 \cdot 10^3) = 39/1$. So in the TS for every 39 video packets one audio packet will be interleaved. We have already seen that the output bit rate of the TS multiplexer is 750 Kbytes/s. So we can put a total of $\lfloor 750 \cdot 10^3 / 188 \rfloor = 3989$ transport packets per second. Among these we can interleave 3960 video and audio packets and the remaining 29 packets can be of private data.

While multiplexing the different elementary streams as long as video, audio, and private data packets are available the multiplexer follows the above ratios to interleave video, audio, and private data packets as mentioned above. If say all the video packets are sent and only audio and private data packets are left to be sent then the multiplexer follows the statistical multiplexing and will give the priority to the audio packets and send them in the bandwidth allocated to the video and audio packets. When all the video and audio packets are sent then all the available bandwidth is utilized for private data packets only.

Example

Consider the above mentioned streams used for model calculations. When these streams are packetized using the TS multiplexer a total of 17694 video, 7687 audio

and 711 private data transport packets are formed. The duration of this generated transport stream is 6.5485 second. So per second the multiplexer puts 3861 video packets, 39 audio packets and 39 private data packets as long as all of video, audio and private data are available. But when all the video packets are transmitted, the bandwidth allocated for video packets is utilized by audio packets. If all the audio packets are also sent then the multiplexer gives all the available bandwidth to the private data streams.

4.3 Software Details

Two programs were written to implement the two levels of the TS multiplexer. The first program *pcspackets.c* generates the PES packets from different input elementary streams. The video PES packets are stored in the file *video.p2v*. The audio PES packets are stored in the file *audio.p2a*. The private data packets are stored in the file *data.p2d*. The second program *tspackets.c* will generate the TS packets by multiplexing the data from different PES packets as mentioned above. In the same program, interprocess communication was set up between two processes using the concept of pipes in UNIX. One process (parent) generates the TS packets and writes them to a file named *tspackets.p2t*. The structure containing the header elements of the generated TS packets will be written to a pipe. This pipe acts as a bridge between the two processes. The second process (child) reads the TS packet header structure from the pipe and checks whether the generated packet contains video or audio or private data as its payload. It will also see for other information like whether adaptation field is present in that packet or not etc. When the first process completes generating the TS packets, the second process writes the following information about the generated TS to the file *STAT.P2T*:

number of video packets

number of audio packets

number of data packets

number of total packets

number of packets containing the adaptation field

The following figure shows the block diagrammatic representation of the above operations.

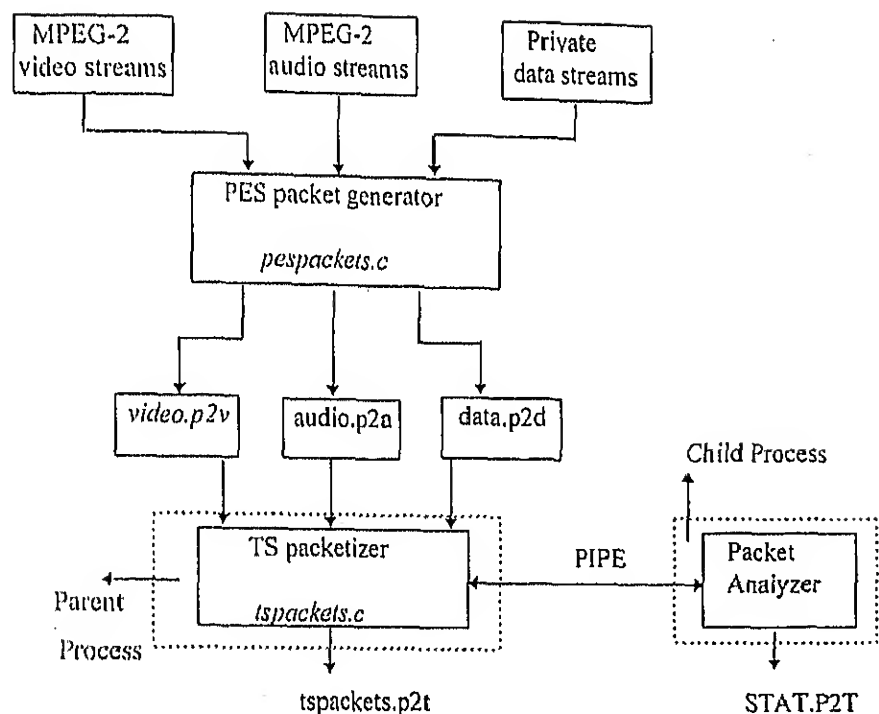


Fig. 4.3 Block diagrammatic representation of TS multiplexer

Table 4.3 shows results obtained when the TS multiplexer was subjected to different video and audio elementary streams and a common private data stream. The number of private data packets is not mentioned in the table as, that number is the same in all the cases and is equal to 741.

Name and size of audio and video streams	Ratio of video to audio bit rates	# of video packets	# of audio packets	duration of the generated TS (seconds)
gi6.m2v (311KB) te01.m2a (36KB)	38:1	1754	204	0.6766
gitape.m2v (933.888KB) te15.m2a (144KB)	39:1	5224	800	1.696
tek7.m2v (825KB) verd16s.m2a (200KB)	195:1	4609	1115	1.6207
hhilong.m2v (3.169MB) genzmer.m2a (1.391MB)	39:1	17694	7687	6.5485
chroma.m2v (37KB) lsf16m.m2a (9KB)	39:1	218	58	0.2549
ccett.m2v (80KB) te26.m2a (15KB)	20:1	457	92	0.3233

lec m2v (311KB)	331	1754	553	0.7640
com_1 m2v (28KB)				
lec m2v (217KB)	151	1356	125	0.557
lec m2a (221KB)				
lec7 m2v (825KB)	1501	4601	451	1.4542
lec03 m2v (80KB)				
lec m2v (217KB)	131	1355	271	0.5333
voidy m2v (17KB)				

Table 4.3 Results

The graph shown in the next page (Fig. 4.3) gives distribution of private data packets in a TS for four different cases. If we consider the case represented by ts2 (pink colored one) this corresponds to the streams which we considered in model calculations. As already mentioned above, there are a total of 17604 video, 7687 audio and 741 private data transport packets in this TS stream. For the first four seconds, 3861 video, 2250 audio and 29 private data packets are present in the TS stream. In the fifth second, there are 2250 video, 1710 audio and 29 private data packets. Note that in the fifth second, audio packets occupied a portion of bandwidth which is unutilized by the video packets. In the sixth second, 3960 audio and 29 private data packets are there in the TS. In the last second, 1621 audio and 507 private data packets are present. So in the graph, for the first six seconds, there are 29 private data packets and in the final second, the number of private data packets rises to 507. Similarly, the other cases can also be interpreted using Table 4.3 and the graph.

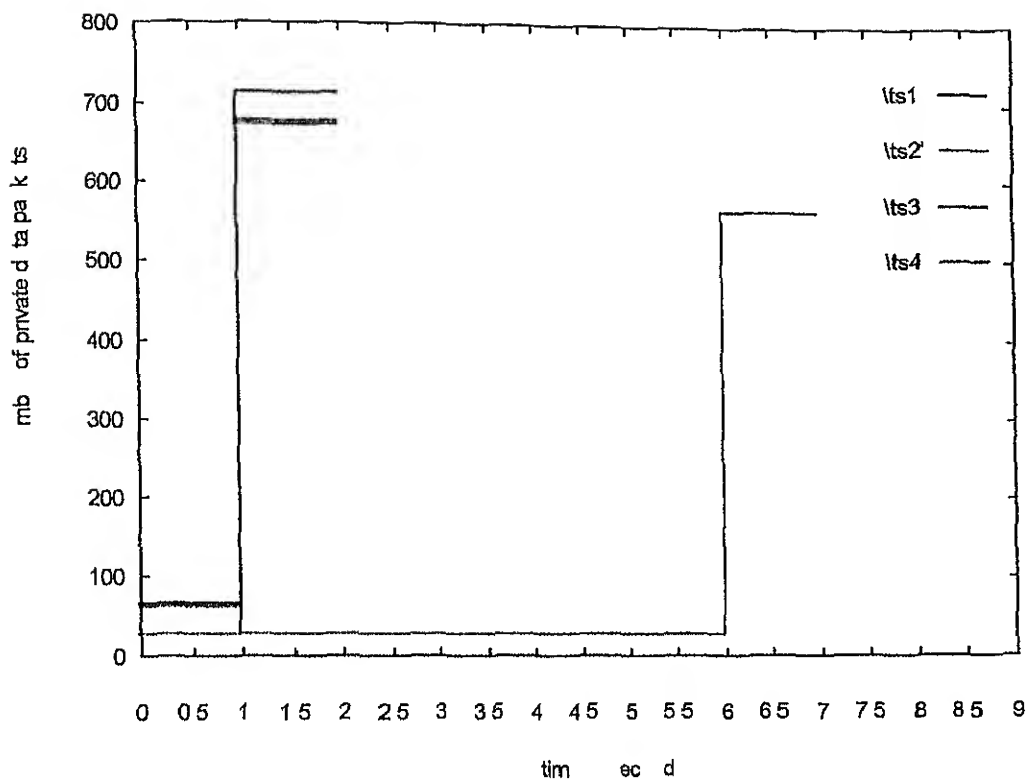


Fig 5.4 Distribution of private data packets in TS

Name of the TS	Name of the component video stream	Name of the component audio stream
ts1	gitape m2v	te15 m2a
ts2	hhilong m2v	genzmer m2a
ts3	tek7 m2v	verd16s m2a
ts4	tek7 m2v	te03 m2a

Conclusions from the above graph

- The private data packets occupy only the un utilized portion of the bandwidth
- Larger the size of the input video and audio elementary streams longer the duration to transmit the private data

Chapter 5

Conclusions and Future Work

5.1 Conclusions

In this work we have studied theoretical and implementational issues involved in the TS multiplexing and implemented a model TS multiplexer. The performance of this multiplexer was studied by subjecting it to different video and audio elementary streams. We found that if the bit rates of the input video elementary streams are higher than 8 Mbit/s (for example HDTV quality video) the multiplexing delay is rather high which in turn causes severe distortion in the picture when the video is being replayed at the receiver's end. This is expected since the rate of the input video elementary stream is higher than the multiplexing rate. In the present implementation the multiplexing rate is fixed as an input parameter to the TS multiplexer. The parameter is fixed in view of the channel BW available. In our case we assumed a channel BW of 6 Mbit/s corresponding to the satellite capacity of 4.375 MHz per channel. In cases where higher quality video has to be transmitted TS multiplexer output rate has to be adjusted accordingly. For the VHS quality video an output bit rate of 6 Mbit/s is good enough and will cause negligible multiplexing delay.

5.2 Future Work

The concept of TS multiplexing can be extended for multiple programs. In our work we have only calculated the amount of private data which can be sent along with video and audio elementary streams. One of the interesting applications of private data transmission is IP over MPEG-2. A model TS multiplexer can be used by implementing an IP layer above the MPEG-2 transport layer.

Part 7 Non Backward Compatible Audio This addresses the need for a new syntax to efficiently de-correlate discrete multichannel surround sound audio. By contrast MPEG 2 audio attempts to code the surround channels as an ancillary data to MPEG 1 backwards compatible Left and Right channels. This allows existing MPEG 1 decoders to parse and decode only the two primary channels while ignoring the side channels.

Part 8 This is a 10 bit video extension. This is an extension of Part 2 and describes the syntax and semantics for coded representation of video with 10 bits of sample precision. The primary application is studio video.

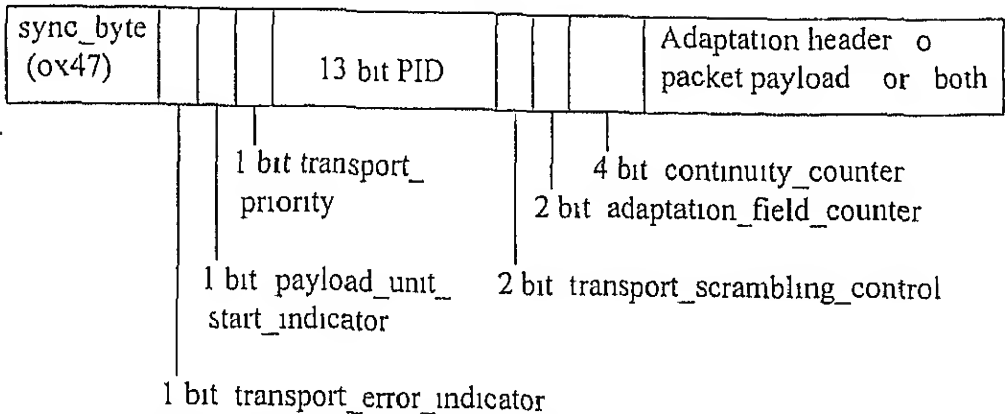
Note: This part has been withdrawn due to lack of interest by industry.

Part 9 Real Time Interface Defines a syntax for video on demand control signals between set top boxes (receivers) and head end servers.

Appendix B

Detailed Description of Various Fields in Transport Packet Header

Semantic definition of fields in Transport header



sync_byte The sync byte is a fixed 8 bit field whose value is 0100 0111 (0x47)

transport_error_indicator The transport_error_indicator is a 1 bit flag. When set to 1 it indicates that at least 1 uncorrectable bit error exists in the associated TS packet

payload_unit_start_indicator The payload_unit_start_indicator is a 1 bit flag which has normative meaning for TS packets that carry PES packets of *PSI (Program Specific Information) data*

When the payload of the TS packet contains PES data, the payload_unit_start_indicator has the following significance. A 1 indicates that the payload of this TS packet will commence with the first byte of a PES packet and a 0 indicates no PES packet shall start in this TS packet. One and only one PES packet may start in any Transport Packet. This also applies to private streams of stream type 6 (which are described under *program map table*).

When the payload of the TS packet contains PSI data, the payload_unit_start_indicator has the following significance. If the TS packet carries the first byte of a PSI section the payload_unit_start_indicator value shall be 1 indicating that the first byte of the payload of this TS packet carries the *pointer_field*. If the TS packet does not carry the first byte of a PSI section the payload_unit_start_indicator value shall be 0 indicating that there is no pointer_field in the payload.

The meaning of this bit for TS packets carrying only private data is not defined.

transport_priority The transport_priority is a 1 bit indicator. When set to 1 it indicates that the associated packet is of greater priority than other packets having the same PID which do not have the bit set to 1. The transport mechanism can use this to prioritize its data within an elementary stream. Depending on the application the priority field may be coded regardless of the PID or within one PID only.

PID The PID is a 13 bit field indicating the type of the data stored in the packet payload. PID value 0x0000 is reserved for the Program Association Table (mentioned later). PID value 0x0001 is reserved for Conditional Access Table (mentioned later). PID values 0x0002-0x000F are reserved. PID value of 0x1FFF is reserved for null packets.

transport_scrambling_control This is a 2 bit field indicating the scrambling of the TS packet payload. The TS packet header including the adaptation field when present shall not be scrambled. In the case of a null packet the value of the *transport_scrambling_control* field shall not be set to 00.

value	Description
00	not scrambled
01	user defined
10	user defined
11	user defined

Table B.1 transport scrambling control values

adaptation_field_control This is a 2 bit field indicating whether this TS packet header is followed by an adaptation field and/or payload.

value	Description
00	reserved for future use
01	no adaptation field payload only
10	adaptation field only no payload
11	adaptation field followed by payload

Table B.2 adaptation field control values

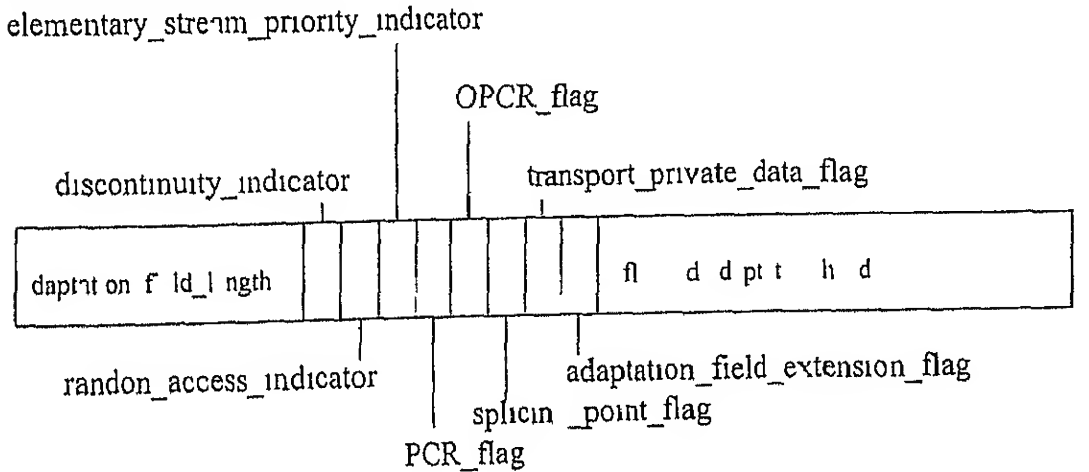
The decoders shall discard TS packets with the *adaptation_field_control* set to a value of 00. In the case of a null packet the value of the *adaptation_field_control* shall be set to 01.

continuity_counter The *continuity_counter* is a 4 bit field incrementing with each TS packet with the same PID. The *continuity_counter* wraps around to 0 after its maximum value. The *continuity_counter* shall not be incremented when the *adaptation_field_control* of the packet equals 00 or 10.

data_byte Data bytes shall be contiguous bytes of data from the PES packets, PSI sections or private data not in these structures as indicated by the PID. In the case of null packets with PID value 0x1FFF data bytes may be assigned any value.

The Adaptation Field

This is a variable length field. Its presence is flagged in the Transport header field with the *adaptation_field_control* field. This field consists of information useful for higher level decoding functions and uses flags to indicate the presence of particular extensions to the field.



Semantic definition of fields in adaptation field

adaptation_field_length The adaptation_field_length is a 16 bit field specifying the number of bytes in the adaptation field immediately following the adaptation_field_length field. The length shall not exceed 18 bytes.

discontinuity_indicator This is a 1 bit field which when set to 1 indicates that the next PCR in a TS packet with the same PID represents a sample of a new system time clock for the associated program. This occurs when bit streams are spliced. This flag should be used at the receiver to change the phase of the clock.

random_access_indicator The random_access_indicator is a 1 bit field. When set to 1 it indicates that the next PES packet in this TS packet or in subsequent TS packets with the same PID shall contain a PTS field and an elementary stream access point. The elementary access points are defined as follows:

Video the first byte of a video sequence header

Audio the first byte of an audio frame

In TS packets where the random_access_indicator is set to a value of 1 there shall be an adaptation field which shall contain at least the program_clock_reference_base and program_clock_reference_extension fields. When set to a value of 0 it is not defined whether a random_access_indicator point occurs or not. The meaning of this flag is not defined for private data.

elementary_stream_priority_indicator This is a single bit flag. It indicates within this PID the priority of the elementary stream data carried within this payload of this TS packet. A 1 indicates that the payload has a higher priority than the payloads of other TS packets. In case of video this field may be set to 1 only if the payload contains one or more bytes from an intra-coded slice. A value of 0 indicates that the payload has the same priority as all other packets which do not have this bit set to a value of 1.

PCR_flag The PCR_flag is a 1 bit flag. A 1 indicates that the adaptation field contains PCR field. A value of 0 indicates that the adaptation field does not contain any PCR fields.

OPCR_flag The OPCR_flag is a 1 bit flag. A 1 indicates that the adaptation field contains an OPCR field. A value of 0 indicates that the adaptation field does not contain any OPCR fields.

splicing_point_flag This is a 1 bit flag. When set to 1 it indicates that a splice_countdown field shall be present in the associated adaptation field specifying the occurrence of a splicing point. A value of 0 indicates that a splice_countdown field is not present in the adaptation field.

transport_private_data_flag This is a 1 bit flag. A value of 1 indicates that the adaptation field contains one or more private data bytes.

adaptation_field_extension_flag This is a 1 bit field which when set to 1 indicates the presence of an adaptation field extension. This feature is included to allow for future growth of adaptation field.

flagged adaptation header fields

PCR_field This is a 42 bit field coded in two parts. Its value is given in the following equation:

$$PCR_base(i) = ((system_clock_frequency - t(i))/300) / \tau$$

$$PCR_ext(i) = ((system_clock_frequency - t(i))) / 300$$

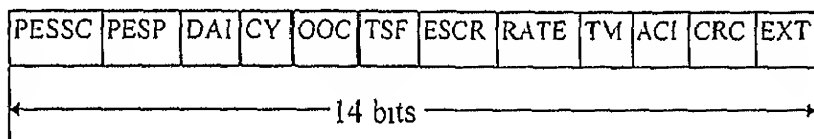
$$PCR(i) = PCR_base(i) * 300 + PCR_ext(i)$$

1111 xxxx	reserved data stream number xxxx 0100 1110 are reserved
1111 1111	program stream directory

Table B 3 stream id table

PES_packet_length A 16 bit field specifying the number of bytes in the PES packet following the last byte of the field. A value of 0 indicates that the PES packet length is neither specified nor bounded and is allowed only in PES packets whose payload is a video elementary stream contained in TS packet.

PES Header Flags These describe the organization of the PES header.



PESSC PES_packet_scrambling_control This 2 bit field indicates the scrambling mode of the PES packet payload.

Value	Description
00	Not scrambled
01	user defined
10	user defined
11	user defined

Table B 4 PES packet scrambling control values

PESP PES_priority This is a 1 bit field indicating the priority of the payload in this PES packet. A 1 indicates a higher priority of the payload of the PES packet payload than a PES packet payload with this field set to 0.

DAI data_alignment_indicator A 1 bit flag which when set to 1 indicates that the PES packet is immediately followed by the access unit data type. The data type is indicated by the *data_stream_alignment_descriptor*.

data_stream_alignment_descriptor This describes which type of alignment is present in the associated elementary stream when the DAI flag is set to 1.

alignment type	Description
00	reserved
01	Slice picture GOP or SEQ
02	picture GOP or SEQ
03	GOP or SEQ
04	SEQ
05 0F	reserved

Table B 5 Video stream alignment values

alignment type	Description
01	reserved
01	Audio Frame
02 FF	reserved

Table B 6 Audio stream alignment values

CY *copyright* If this bit is set to 1 it indicates that the contents of the associated PES packet payload is copyrighted

OOC *original_or_copy* If this bit is set to 1 the contents of the associated PES packet payload is an original. When set to 0 it indicates that it is a copy

TSF *PTS_DTS_flags* This is a 2 bit flag. If its value is 10 a PTS field is present in the PES packet header. If its value is 11 both a PTS and DTS field are present in the PES packet header. If its value is 00 neither a PTS nor a DTS field is present in the PES packet header. The value 01 is forbidden.

ESCR *ESCR_flag* A 1 bit flag when set to 1 indicates that an ESCR field is present in the PES packet header.

RATE *ES_rate_flag* A 1 bit flag when set to 1 indicates that the ES_rate field is present in the PES packet header.

TM *DSM_trick_mode_flag* A 1 bit field when set to 1 indicates the presence of an 8 bit field used for trick_mode_control field. It indicates the presence of an 8 bit field describing the DSM (Digital Storage Media) operating mode.

ACI *additional_copy_info_flag* A 1 bit flag when set to 1 indicates the presence of the additional_copy_info field.

CRC *PES_CRC_flag* A 1 bit flag when set to 1 indicates that a CRC field is present in the PES packet.

LXT *PES_extension_flag* A 1 bit flag when set to 1 indicates that an extension field of flags exists in this PES packet header.

PES_header_data_length An 8 bit field specifying the total number of bytes occupied by the optional fields and any stuffing bytes contained in this PES packet header. The presence of optional fields is indicated in the byte that precedes the PES_header_data_length field.

PES Header Fields

PIS DTS	ESCR	ES t	DSM t k n d	add t py f f ld	PES t fl g	t _d t f ld	stuff n_ byt
------------	------	------	----------------	--------------------	---------------	----------------	-----------------

PTS (*presentation_time_stamp*) PTS informs the decoder of the intended time of presentation of presentation unit that corresponds to the first access unit that commences in the packet. The value of PTS is measured in the number of periods of a 90KHz system clock.

DTS (*decoding_time_stamp*) DTS informs the intended time of decoding of an access unit. This is also a 33 bit number coded in three separate fields.

ESCR The elementary_stream_clock_reference (ESCR) is an optional 47 bit field coded as a 3 bit base field with a 9 bit extension field. It indicates the intended time of arrival of the byte containing the last bit of the base field at the input of the system target decoder when present in PES Stream.

ESCR_rate The elementary_stream_rate field is a positive integer specifying the rate at which the system target decoder receives bytes of the PES packet in the case of PES stream. This field is valid in the PES packet in which it is included and in

subsequent PES packets of the same PES stream until a new ES_rate field is encountered. The value of the ES_rate is measured in units of 50 bytes/second rounded up values.

DSM_trick_mode This is an 8 bit field indicating the nature of the information encoded. This field is further partitioned as follows. The field is further partitioned as follows:

trick_mode_control (3 bits)

field_id (2 bits)

intra_slice_refresh (1 bit) and

frequency_truncation (2 bits)

trick_mode_control It indicates nature of the DSM Mode

000 Fast Forward

001 Slow Motion

010 Freeze Frame

011 Fast reverse

1xx Reserved

field_id This identifier is valid interlaced pictures only and describes how the current frame is to be displayed

00 Display field 1 only

01 Display field 2 only

10 Display complete frame

11 Reserved

intra_slice_refresh This field indicates that each picture is composed of intra slices with possible gaps between them. The decoder should replace the missing slices by repeating the collocated sites from the previously decoded picture.

frequency_truncation This field indicates the selection of coefficients from the DSM

00 only DC coefficients are sent

01 The first three coefficients in scan order on average

10 The first six coefficients in scan order on average

This field is for information purposes only. At times more than the specified number of coefficients may be sent. At other times less than the specified number of coefficients may be sent.

PES Extension Flags

PES packet flag	picture header flag	program packet flag	STDB flag	Reserved	PES extension field flag
-----------------	---------------------	---------------------	-----------	----------	--------------------------

The PES header can contain additional flags if the PES Extension Flag is set. These flags are transmitted in a one byte data field as shown above. The flags indicate whether further extension to the PES header exist. In each case the flag is set to 1 if the header field is present.

PES private data flag A 1 bit flag when set to 1 indicates that the PES packet header contains private data.

packet header field flag A 1 bit flag when set to 1 indicates that an ISO/IEC 11172 packet header or an ISO/IEC 13818 Program Stream packet header is stored in this PES packet header. If this field is in a PES packet that is contained in Program Stream then this field shall be set to 0. In a TS when set to a value of 0 it indicates that no packet header is present in the PES header.

program_packet_sequence_counter_flag A 1 bit flag when set to 1 indicates that this field is present in this PES packet

STD_buffer_flag A 1 bit flag when set to 1 indicates that the STD_buffer_scale and STD_buffer_size are available in the PES packet header

PES_extension_flag This is a 1 bit field when set to 1 indicates the presence of the PES_extension_field

PES_private_data This is a 16 byte field which contains private data. This data combined with the fields before and after shall not emulate the packet_start_code_prefix (0x000001)

packet_field_length This is a 8 bit field which indicates the length in bytes of the packet_header_field()

program_packet_sequence_counter This is an 87 bit field. It is an optional counter that increments with each successive PES packet in program multiplex. This allows an application to retrieve the original PES packet sequence of Program Stream. This counter will wrap around to 0 after its maximum value.

stuffing_byte This is a fixed 8 bit value equal to 1111 1111 that can be inserted by the encoder for example to meet the requirements of the digital storage medium. It is discarded by the decoder. No more than 2 stuffing bytes shall be present in a PES packet header.

STD_buffer_scale (Program Stream only) This is a 1 bit field that indicates the scaling factor used to interpret the subsequent STD_buffer_size field. If the preceding stream_id indicates an audio stream STD_buffer_scale shall have the value 0. If the preceding stream_id indicates a video stream this shall have a value 1. For all other stream types the value may be either 1 or 0.

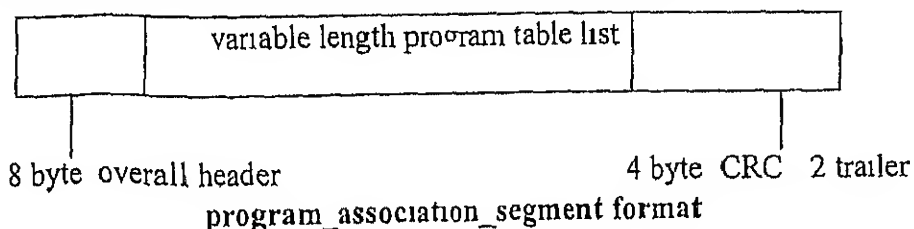
STD_buffer_size (Program Stream only) This is a 13 bit unsigned integer defining the size of the input buffer in STD. If STD_buffer_scale has the value 0 then the buffer_size measures the buffer size in units of 128 bytes. If STD_buffer_scale has the value 1 then the buffer_size measures the buffer size in units of 1024 bytes.

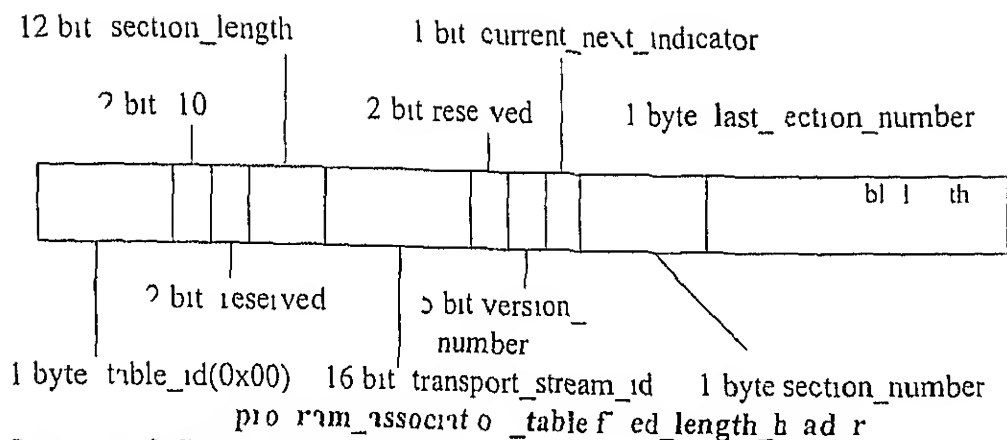
PES_extension_field_length This 7 bit field specifies the length in bytes of the data following this in the PES_extension_field.

PES_packet_data_byte PES_packet_data_bytes shall be contiguous bytes of data from the elementary stream indicated by the packet's stream_id or PID. The byte order of the elementary stream shall be preserved.

padding_byte This is a fixed 8 bit value equal to 1111 1111 that can be inserted by the encoder to meet the requirements of DSM. It is discarded by the decoder.

Program Association Segment and Table Header Formats





Semantic definition of fields in Program Association Table Segment header

pointer_field This is an 8 bit field whose presence in the beginning of the payload portion of the TS packet indicates that the payload contains program association table. The value of this 8 bit field indicates the number of bytes immediately following the pointer_field until the first segment that is present in the payload of the TS packet. When at least one section begins in a given TS packet then the payload_unit_start_indicator shall be set to 1 and the first byte of the payload of that TS packet shall contain the pointer.

table_id This is a 1 byte field which is present common in program association table header, program map table header and conditional access table header and whose value depends on the table in which it is being used.

value	description
0x00	program association table segment
0x01	conditional access table segment
0x02	program map table segment
0x03-0x3F	reserved
0x40-0xFF	user defined
0xFF	forbidden

Table B.7 table_id assignment values

section_length This is a twelve bit field the first two bits of which shall be 00. It specifies the number of bytes of the section starting immediately following the section_length field and including CRC.

transport_stream_id This is a 16 bit field which serves as a label to identify this TS from any other multiplex within a network. Its value is defined by the user.

version_number This 5 bit field is the version number of the whole program association table. The version number shall be incremented by 1 whenever the definition of the Program Association Table changes. Upon reaching the value 1 it wraps around to 0. When the current_next_indicator is set to 1 then the version_number shall be that of the currently applicable program association table. When the current_next_indicator is set to 0 then the version_number shall be that of the next applicable program association table.

current_next_indicator A 1 bit indicator which when set to 1 indicates that the program association table is sent is currently applicable. When the bit is set to 0 it indicates that the table sent is not yet applicable and shall be the next table to become valid.

section_number This 8 bit field gives the number of this section. The *section_number* of the first section in the program association table shall be 0x00. It shall be incremented by 1 with each additional section in the program association table.

last_section_number This 8 bit field specifies the number of the last section of the complete program association table.

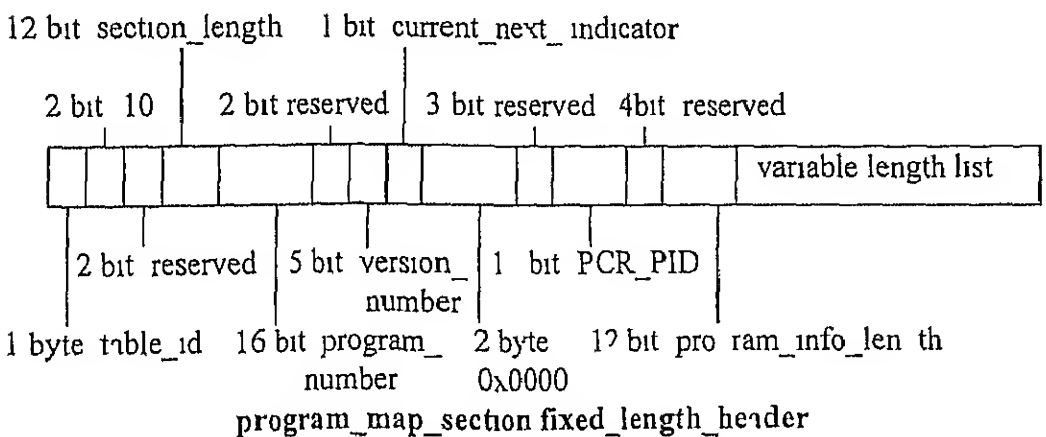
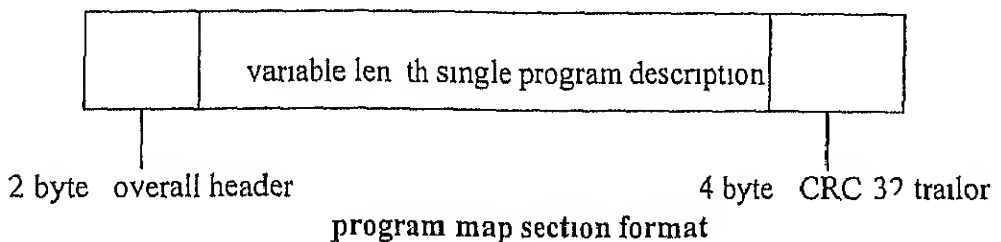
variable length fields

program_number This is a 16 bit field. It specifies the program to which the *program_map_PID* is applicable. If this is set to 0x0000 then the following PID reference shall be the network PID. For all other cases the value of this field is user defined. This field shall not take any single value more than once within one version of the program association table.

network_PID *Network_PID* is a 13 bit field specifying the PID of the TS packet which shall contain the Network Information Table. The value of the *network_PID* field is defined by the user but shall not take values reserved for other purposes. The presence of the *network_PID* is optional.

program_map_PID This is a 13 bit field specifying the PID of the TS packet which shall contain the *program_map_section* applicable for the program as specified by the *program_number*. No *program_number* shall have more than one *program_map_PID* assignment. The value of the *program_map_PID* is defined by the user but shall not take the values reserved for other purposes.

Program Map Table Details



Semantic definition of fields in TS program map section

table_id This will always be set to 0x02.

section_length This is a twelve bit field the first two bits of which shall be 00. It specifies the number of bytes of the section starting immediately following the *section_length* field and including CRC.

program_number This is a 16 bit field. It specifies the program to which the *program_map_PID* is applicable. One program definition shall be carried within only one *program_map_section*. So program definition is never longer than 1016 bytes.

version_number Its function is same as that in program association table.

current_next_indicator Its function is also identical to that of in program association table.

PCR_PID This is a 13 bit field indicating the PID of the TS packet which shall contain the PCR fields valid for the program specified by *program_number*. If no PCR is associated with a program definition for private streams then this field shall take the value of 0x1FFF.

program_info_length This is a twelve bit field, the first two bits of which shall be 00. It specifies the number of the descriptors immediately following this field.

stream_type This is an 8 bit field specifying the type of elementary stream or payload carried within the packets with the PID whose value is specified by the *elementary_PID*. The values of *stream_type* are specified in the following table.

Value	Description
0x00	reserved
0x01	MPEG 1 video
0x02	MPEG 2 video
0x03	MPEG 1 audio
0x04	MPEG 2 audio
0x05	MPEG 2 private sections
0x06	MPEG 2 PES packets containing private data
0x07	ISO/IEC 13522 MHEG
0x08	ISO/IEC 13818 DSM-CC
0x09	MPEG 2 private data
0x6A-0x7F	MPEG 2 reserved
0x48-0xFF	user defined

Table B.8 stream type assignment table

elementary_PID This is a 13 bit field specifying the PID of the Transport packets which carry the associated elementary stream or payload.

ES_info_length This is a twelve bit field, the first two bits of which shall be 00. It specifies the number of bytes of the descriptors of the associated elementary stream immediately following this field.

Appendix C

The DVB Standards

The DVB Project Office has so far published the following standards as European Telecommunication Standards Institute (ETSI) standards

DVB S Digital broadcasting systems for television sound and data services
Framing structure channel coding and modulation for 11/12 GHz satellite services
ETS 300 421 December 1994

The modern standard for satellite broadcasts with various data rates BW requirements and error correction capabilities

DVB C Digital broadcasting systems for television sound and data services
Framing structure channel coding and modulation for cable systems ETS 300 429
December 1994

The modern standard for cable broadcasts with various data rates levels of noise immunity and BW requirements

DVB CS Digital broadcasting systems for television sound and data services
Satellite Master Antenna Television (SMATV) distribution systems ETS 473 May 1995

Description of the alternative ways in which a community sat antenna installation head end could operate

DVB SI Digital broadcasting systems for television sound and data services
Specification for Service Information (SI) in Digital Video Broadcasting (DVB) systems ETS 300 468 October 1995

Electronic program guide and automatic tuning and VCR control information broadcasted in DVB data streams

DVB TXT Digital broadcasting systems for television sound and data services
Specification for conveying ITU R System B Teletext in Digital Video Broadcasting (DVB) bitstreams ETS 300 472 May 1995

MPEG 2 encapsulation for the European analog teletext data

DVB CI Common Interface Specification for Conditional Access and other Digital Video Broadcasting Decoder Application European Standard prEN 50221 Draft D 1996 04 23

PCMCIA slot that will allow to add proprietary access control and other extensions

The DVB S standard has now also been included in ITU R recommendations
Further DVB standards for terrestrial digital broadcast (DVB T) subtitling MMDS and interactive TV are under preparation

In addition two ETSI Technical Reports (ETR) have been published that are not formal standards but form important guidelines for systems implementors

ETR 154 Digital broadcasting systems for television Implementation guidelines for the use of MPEG 2 systems Video and audio in satellite and cable broadcasting applications ETSI Technical Report draft ETR 154 1994 11

The MPEG_2 standard ISO/IEC 13818 specifies a very large number of system parameters and optional features This text specifies many MPEG minimum parameter ranges and options which every DVB receiver is supposed to support

ETR 211 Digital broadcasting systems for television Implementation guidelines for the use of MPEG 2 systems Guideline on implementation and use of service information ETSI Technical Report final draft ETR 211 1996 02 12

This text provides many important details for implementing the DVB SI standard

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- 5 Radio Communications Study Groups *A Guide to Digital Terrestrial Television Broadcasting in the VHF/UHF Bands* ITU Documnet 11/4 E 1996
- 6 Christos Tryfonas *MPEG 2 Transport Over ATM Networks* M S Thesis University of California Santa Cruz 1996
- 7 Barry G Haskell Atul Puri and Arun N Netravali *Digital Video An Introduction to MPEG 2* Chapman & Hall 1997
- 8 Bruce R Elbert *The Satellite Communication Applications Handbook* Artech House 1997
- 9 Daniel Minoli Robert Kemath *Distributed Multimedia Through Broadband Communications Services* Artech House 1994
- 10 Lecture Notes *Short Course on Digital Broadcasting* IIT Kanpur 1997
- 11 Sandhya V Sule *Report on Development of Interactive Multimedia Information Services over a Hybrid Internet and Broadcast Digital TV Network* IIT Kanpur 1997

- 12 Peter Noll MPEG Digital Audio Coding *IEEE Signal Processing Magazine* pg 59-80 September 1997
- 13 Web site for MPEG 2 Video Streams <http://www.mpeg2.de/video/streams>
- 14 Web site for MPEG 2 Software Video Encoders and Decoders
<http://www.mpeg2.de/video/software.htm>
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<http://www.mpeg2.de/audio/streams>
- 16 Web site for MPEG 2 Software Audio Encoders and Decoders
<http://www.mpeg2.de/audio/software>
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- 18 W. Richard Stevens *UNIX Network Programming* PHI 1990
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